MultiVOIP™

Voice/Fax over IP Networks

Voice/Fax Over IP Networks
Model MVP400

H.323 Mode

User Guide

MultiTech® Systems

User Guide

S0000008 Revision C

MultiVOIP 400 (Model MVP400)

This publication may not be reproduced, in whole or in part, without prior expressed written permission from Multi-Tech Systems, Inc. All rights reserved.

Copyright © 2001 by Multi-Tech Systems, Inc.

Multi-Tech Systems, Inc. makes no representations or warranties with respect to the contents hereof and specifically disclaims any implied warranties of merchantability or fitness for any particular purpose. Furthermore, Multi-Tech Systems, Inc. reserves the right to revise this publication and to make changes from time to time in the content hereof without obligation of Multi-Tech Systems, Inc. to notify any person or organization of such revisions or changes.

Record of Revisions

Revision	Description
A (6/5/00)	Added H.323 protocol support; covers software version 3.50. All pages at revision A.
B (1/24/01)	Manual updated for software version 3.51.
C (12/26/01)	Removed references to MultiVOIP 800

Patents

This Product is covered by the following U.S. Patent Numbers: ***6151333, 5757801, 5682386***. Other Patents Pending.

TRADEMARK

Multi-Tech and the Multi-Tech logo are registered trademarks and MultiVOIP is a trademark of Multi-Tech Systems, Inc.

Adobe Acrobat is a trademark of Adobe Systems Incorporated.
Microsoft Windows, Windows 2000, Windows 98, Windows 95, Windows NT, and NetMeeting are either registered trademarks or trademarks of Microsoft Corporation in the United States and/or other countries.

Multi-Tech Systems, Inc.
2205 Woodale Drive
Mounds View, Minnesota 55112
(763) 785-3500 or (800) 328-9717
Fax 763-785-9874
Technical Support (800) 972-2439
Internet Address: **http://www.multitech.com**

Contents

Chapter 1 - Introduction and Description 5

Introduction 6
Preview of this Guide 7
Typical Application 8
Front Panel Description 13
Back Panel Description 14
  Power Connector 14
  Command Port Connector 14
  10Base-T (Ethernet) Connector 14
  Voice/Fax Channel 14
Specifications 15
  Ethernet Port 15
  Command Port 15
  Voice/Fax Channel 15
  Electrical/Physical 15

Chapter 2 - Installation 17

Installing Your MultiVOIP 18
  Installing and Configuring Your MultiVOIP 18
  Deploying the VOIP Network 18
Safety Warning Telecom 18
Unpacking Your MultiVOIP 19
Safety Warnings 19
Cabling Procedure 20
  E&M Jumper Block Positioning Procedure 21

Chapter 3 - Software Loading and Configuration 23

Loading Your MultiVOIP Software 24
Configuring Your MultiVOIP 400 27
  Registering with a Gatekeeper Phone Directory 33
  Building a Proprietary Phonebook Directory 38
  Configuring Your Slave MultiVOIP 400s 46
Deploying the VOIP Network 54
  Remote Site Administrator 54

Chapter 4 - MultiVOIP 400 Software 57

Introduction 58
  Before You Begin 58
MultiVOIP 400 Configuration 59
Changing Channel Parameters 60
  Interface 60
  Voice/Fax 62
  Billing/Misc 63
  Regional 65
Changing the Phone Directory Database 66
  Proprietary Phone Directory Database 67
  Gatekeeper Phone Directory Database 70
Changing IP Parameters 73
Viewing Call Progress 74
Applications Setup 75

Chapter 5 - Remote Configuration and Management 77

Viewing Statistics 78
IP Statistics 78
SNMP Statistics 79
Viewing Logs 80
Viewing Log Entry Details 80
Viewing Channel Totals 81
Reports 81
Upgrade Procedures 82
Download Firmware 82
Manual Upgrade Procedure 84
Introduction 85
Remote Configuration 85
Modem-Based 85
LAN-Based 87
Remote Management 88
Telnet 88
WEB Management 90

Chapter 6 - Warranty, Service and Tech Support 93

Introduction 94
Limited Warranty 94
Addendum for North American Products 94
Addendum for International Products 94
Service 95
Ordering Accessories 95
Tech Support 96
Recording MultiVOIP 400 Information 96
About the Internet 96

Appendixes 97

Appendix A - TCP/IP Description 98
Appendix B - Cabling Diagrams 101
Appendix C - Regulatory Information 103
Class A Statement 103
Fax Branding Statement 103
FCC Part 68 Telecom 104
Canadian Limitations Notice 105
EMC, Safety and Terminal Directive Compliance 105

Glossary 107

Index 120

MultiVOIP™

Voice/Fax over IP Networks

Chapter 1 - Introduction and Description

MultiTech® Systems

Introduction

Welcome to Multi-Tech's new voice/fax gateway, the MultiVOIP, model MVP400. The MultiVOIP 400 enables analog voice and fax communication over a traditional data communications/data networking digital Internet/Intranet. Multi-Tech's new voice/fax gateway technology allows voice and fax communication to be transmitted, with no additional expense, over your existing communications Internet/Intranet, which has traditionally been data-only. To access this free voice and fax communication, all you have to do is connect the MultiVOIP 400 to a phone or to your existing in-house phone switch and then to your existing Internet/Intranet connection. Once configured, the MultiVOIP 400 allows voice and fax to travel down the same path as your traditional data communications.

The MultiVOIP 400 supports the H.323 standards-based protocol enabling your MultiVOIP 400 to participate in real-time conferencing with other third-party VOIP Gateways or endpoints that support the H.323 protocol (for example, Microsoft NetMeeting®). The H.323 standard defines how endpoints make and receive calls, how endpoints negotiate a common set of audio and data capabilities, how information is formatted and sent over the network, and how endpoints communicate with their respective Gatekeepers. Gatekeeper software is optional and when used in a network, it typically resides on a designated PC. It acts as the central point for all calls within its zone and provides call control services to all registered endpoints. In addition, Gatekeepers can perform bandwidth management through support for Bandwidth Request, Confirm, and Reject messages.

The MVP400 has four voice/fax channels (with three voice/fax interfaces per channel), a 10 Mbps Ethernet LAN interface, and a command port for configuration.

System management is provided through the command port using bundled Windows® software which provides easy-to-use configuration menus and a comprehensive online Help system.

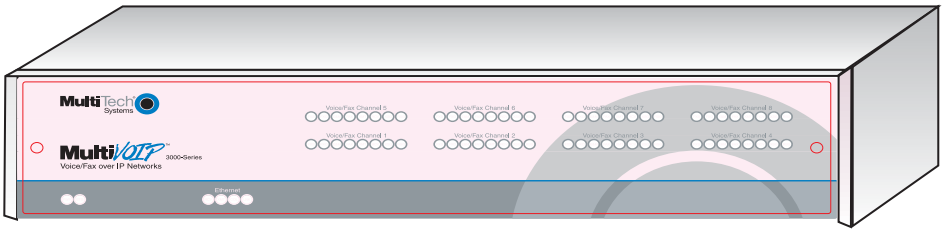


Figure 1-1. MultiVOIP 400

Preview of this Guide

This guide describes the MultiVOIP and tells you how to install and configure the unit. The information contained in each chapter is as follows:

Chapter 1 - Introduction and Description

Chapter 1 describes the MultiVOIP and provides a typical application. Front panel indicator and back panel connector descriptions are provided. In addition, a list of relevant specifications is provided at the end of the chapter.

Chapter 2 - Installation

Chapter 2 provides information on unpacking and cabling your MultiVOIP. The installation procedure describes each cable connection.

Chapter 3 - Software Loading and Configuration

Chapter 3 provides instructions for software loading and initial configuration. The MultiVOIP software CD is Windows® based. Later chapters, as well as the Help, describe the MultiVOIP software in more detail.

Chapter 4 - MultiVOIP Software

Chapter 4 describes the MultiVOIP software package designed for the Windows ® environment. For explanations and parameters of each element within a dialog box, refer to the Help.

Chapter 5 - Remote Configuration and Management

Chapter 5 provides procedures for changing the configuration of a remote MultiVOIP. Remote configuration enables you to change the configuration of a unit by simply connecting two modems between the two MultiVOIPs and remotely controlling the unit. Chapter 5 also describes typical client applications such as Telnet and Web-based management used for remote configuration of the MultiVOIP.

Chapter 6 - Warranty, Service and Tech Support

Chapter 6 provides instructions on getting service for your MultiVOIP at the factory, a statement of the limited warranty, information about our Internet presence, and space for recording information about your MultiVOIP prior to calling Multi-Tech Technical Support.

Typical Application

Before Voice Over IP (VOIP) (voice over the Internet), a corporate office had a data connection to the Internet and a voice connection to the public switched telephone network (PSTN). With VOIP, the two networks can be tied together. To accomplish this, a MultiVOIP is connected between the public switched telephone network and the data network as shown in the typical VOIP application in Figure 1-2. A MultiVOIP 400 at the corporate office is tied into the public switched telephone network thru the corporate PBX and to the data network. The corporate MultiVOIP is an MVP400 with four voice/fax channels connected to the PBX enabling the sales, marketing and regional offices to simultaneously call in or fax on any one of the corporate telephone numbers. The sales office is using a MultiVOIP MVP200 that is connected to the sales office Key Telephone System (KTS) in which a sales attendant answers incoming calls and directs them to their proper destination. The marketing office and regional offices also use MVP200s. The marketing office is set up with the MultiVOIP connected to the E&M trunks on the PBX. The regional office uses an FXS interface to connect two analog telephones directly to the local telephone network.

In this configuration, the corporate MultiVOIP is connected to analog trunks on the corporate PBX and the Ethernet connection is plugged into the hub on the data network. The data network is connected via a router to the Internet. In our typical application, any user at the corporate office can pick up a telephone and call the sales, marketing, or regional office. To accomplish this, the caller picks up the phone and dials a trunk extension, for example extension 4. This is the same principle as dialing 9 for an outside line in a corporate environment. A second dial tone is heard, the caller then dials extension 201 at the sales office. This rings the KTS at the sales office. The sales receptionist answers the call and then directs the call (for example, you wanted to talk to the person at extension 5125) and a voice conservation takes place.

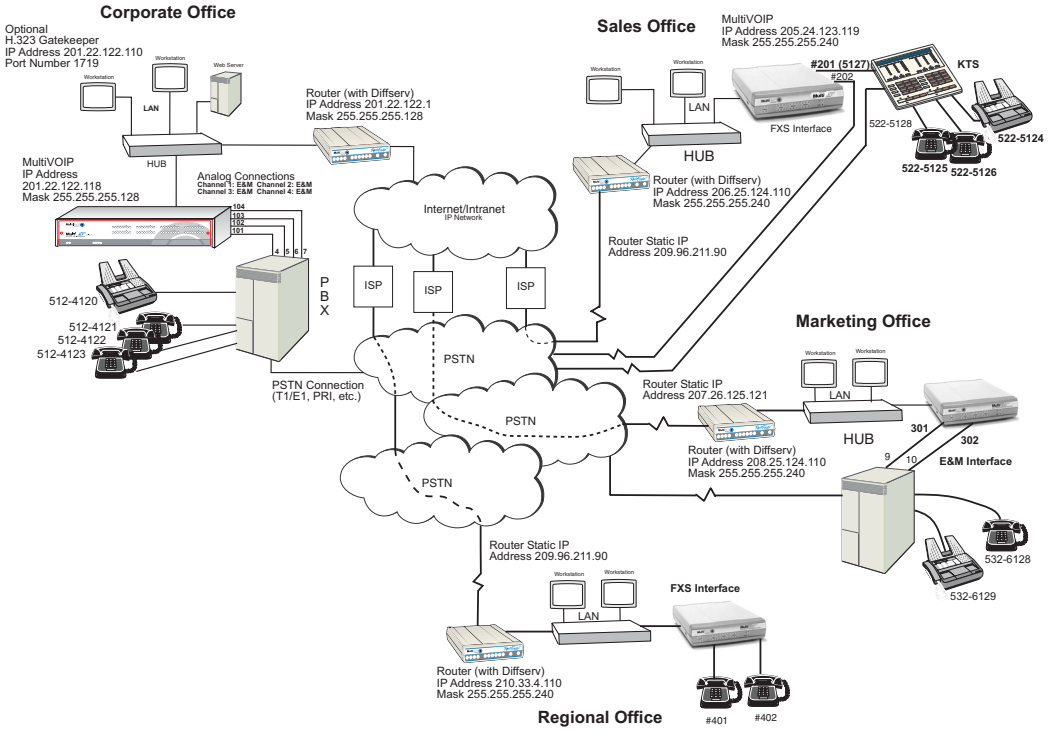


Figure 1-2. Example of a MultiVOIP application

The sales office is set up with a two voice/fax channel MultiVOIP MVP200 connected to the LAN with one voice/fax channel connected directly to the local Public Switch Telephone Network (PSTN) and the second voice/fax channel connected to the trunk side of the KTS. This enables an attendant to direct incoming calls to any one of the telephones or fax machines connected to the Key Telephone System at the sales office. This enables the corporate caller to call either the sales office or instead of dialing extension 201, dials extension 202 and is now connected to the local telephone network at the

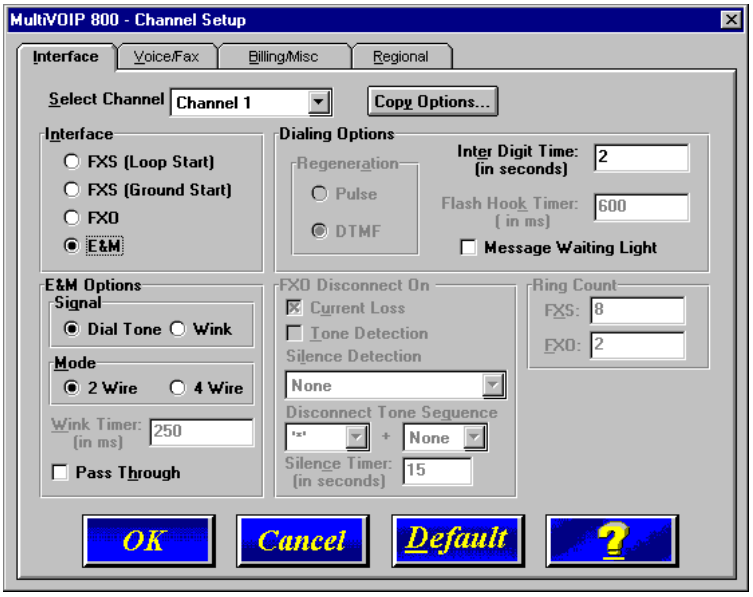
sales office. The corporate caller could then dial any telephone number from the sales office PSTN and this would be a local call.

The regional office is set up very similar to the corporate office with the exception that a MVP200 is connected between the network and the PBX. This enables two voice/fax channels to be used at the regional office. A person in the regional office can pick up a telephone and dial a trunk extension (for example, trunk extension 10) on the local PBX. This is the same as dialing a 9 for an outside line. A second dial tone is heard, the caller would then dial the corporate MultiVOIP at one of the following extensions (101 thru 104). When the third dial tone is heard, the caller could then dial any telephone at the corporate office by dialing its extension, for example, extension 4123.

To configure a MultiVOIP, the COM port of a PC is connected to the Command port on the MultiVOIP. Configuration software is loaded onto your PC and your unique LAN parameters must be established. The configuration software is based on a standard Windows Graphical User Interface (GUI) which simplifies your selection process to a single parameter group within a dialog box. For example, your LAN IP parameters are contained in a single dialog box as shown below. You can configure your network IP address and mask for the MultiVOIP 400 and the gateway address for the corporate router on the same dialog box.

For your corporate MultiVOIP, the Ethernet Frame Type is Type II, the IP Address is 201.22.122.118, the Subnet Mask Address is 255.255.255.128, and the router Gateway Address is 201.22.122.1. Once the LAN parameters are established, you can set up the voice channel parameters.

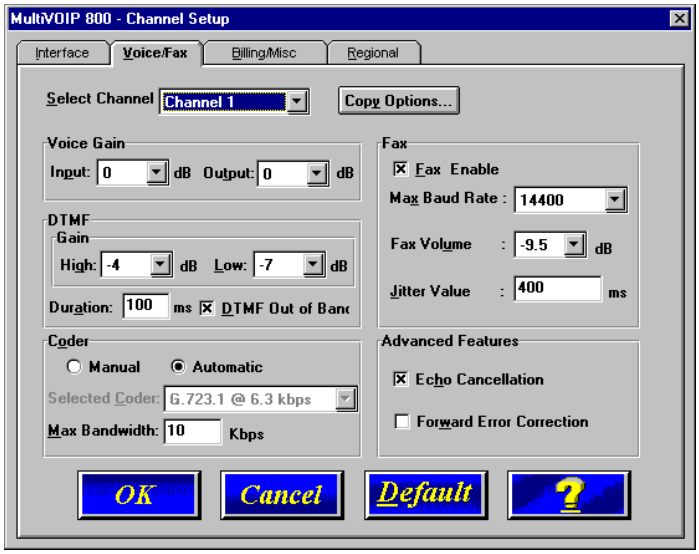
The channel setup parameters define the voice side of the MultiVOIP, the voice channel interface. FXS (Ground and Loop Start) are for connecting to a standard analog telephone set, FXO (Foreign Exchange Office) interface connects to the station side of a PBX, and E&M (Ear and Mouth) connects to the trunk side of the PBX. Along with each interface there are additional parameters that need to be considered, such as for FXO, the dialing options for DTMF (Touch-Tone) or Pulse, the method of disconnecting (Current Loss or Tone Detection), and for E&M, signaling, mode, and the wink timer settings in milliseconds.



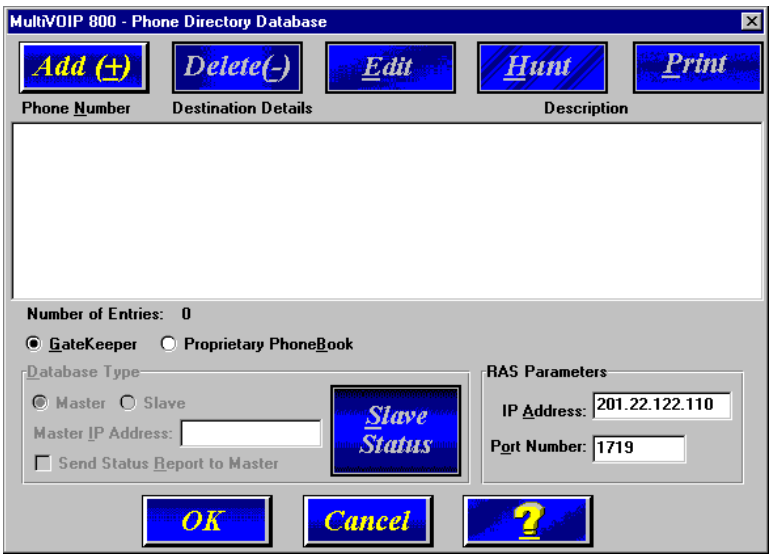
In our typical application, you would configure the corporate office channel parameters for an E&M interface. With this interface, you would have to consult with your in-house telephone people to determine how your PBX is configured. They would have to determine E&M signaling options (dial

tone or wink), 2-wire or 4-wire connection, and if the wink option is the signaling option, the wink timer delay in milliseconds. The default is 250 milliseconds.

Additional channel setup parameters cover the voice coder, DTMF gain, voice gain, and faxing in the Voice/Fax tab of the Channel Setup dialog box. The most important parameter is this group is to ensure that the voice coder is the same for all MultiVOIPs in the network. The **Billing/Misc** tab handles the billing options, automatic disconnect options, and the dynamic jitter buffer options. The jitter options in this tab handle voice break up which can be particularly disruptive to voice communications. For the most part, these parameters can remain in their default values. The **Regional** tab defines the country or region in which the MultiVOIP is being used.



Once you have completed channel setup, you will need to add the phone numbers to the phone directory database. Before you set up the phone directory database, consider how the database is going to be used: will you have an H.323 Gatekeeper set up your call sessions, or are you going to control your call sessions using the proprietary phone book? The H.323 Gatekeeper acts as the central point for all calls within its zone and provides call control services to registered endpoints. If you choose the proprietary phone book, you establish a master-slave relationship where the master MultiVOIP maintains the phone directory and downloads the directory to each slave unit.



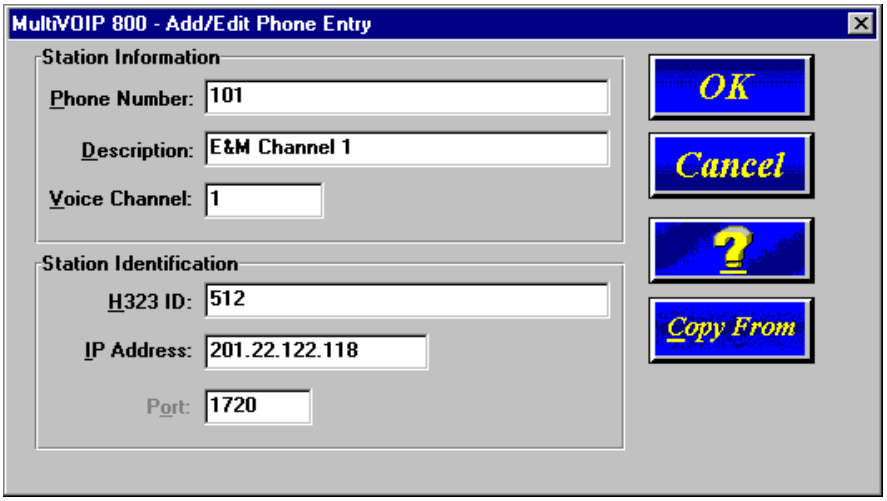
The decision on building the phone directory database is contained in the Phone Directory Database dialog box. Before you choose how the data base is going to be used, here are a couple of things to keep in mind: (1) If a Gatekeeper is employed in the network, you need to choose the Gatekeeper option. You can not mix the Proprietary PhoneBook with the Gatekeeper. If you choose the

Gatekeeper option, you can communicate with other third party endpoints that support H.323 (for example, Microsoft NetMeeting). (2) If you choose the Proprietary PhoneBook, you establish a master-slave relationship in that the master MultiVOIP maintains the phone directory database. All of the phone numbers are listed in the data base so that if you want to communicate with someone in your VOIP network, you can see the phone number in your data base. Every time you bring up your slave MultiVOIP the current phone directory is downloaded from the master MultiVOIP.

The Gatekeeper is a separate application that can operate on a network PC and provides all the controls needed to create, control, and manage an H.323 network zone. The H.323 network zone is all the endpoints (terminals and gateways (MultiVOIPs)) that register with the gatekeeper. The gatekeeper functions are address translation from LAN aliases for terminals and gateways to IP addresses as defined in the RAS (Registration/Admission/Status) specification. The RAS Protocol defines the communication with a gatekeeper and support for RTP/RTCP for sequencing audio packets. The H.323 Gatekeeper also provides call-authorization for both accepting and placing calls in its zone, and certain monitoring features such as call permissioning and address resolution.

So, if you choose the **Gatekeeper** option, initially you need to communicate with the administrator of the Gatekeeper to preregister your MultiVOIP. The information you need from the Gatekeeper administrator is the IP address of the Gatekeeper and its port number. Then you need to establish your alias address which includes phone number, channel number, H323 ID, and your MultiVOIP LAN IP address. The port number is 1720, but if the Gatekeeper uses a different port number, you have to ensure that you use the corrsponding pair. The Gatekeeper administrator will then enter your information into the Gatekeeper data base. This concludes the preregistration.

Now, you can enter your alias address information into the **Add/Edit Phone Entry** dialog box. For example, if you were setting up the corporate MultiVOIP, you could enter the following information for Voice Channel 1. For instance, in our typical application channel 1 of the Corporate MultiVOIP uses extension 101. The Description is optional, but can be helpful if it is assigned to a particular individual or department, or in this case it defines the channel interface. The H323 ID is an alias that is sent to the Gatekeeper. The H323 ID can be a series of numbers or letters, but if you have a NetMeeting end point, it would be better to use a series of numbers. The IP Address of the Corporate MultiVOIP is 201.022.122.118 and the default port number 1720 is used.

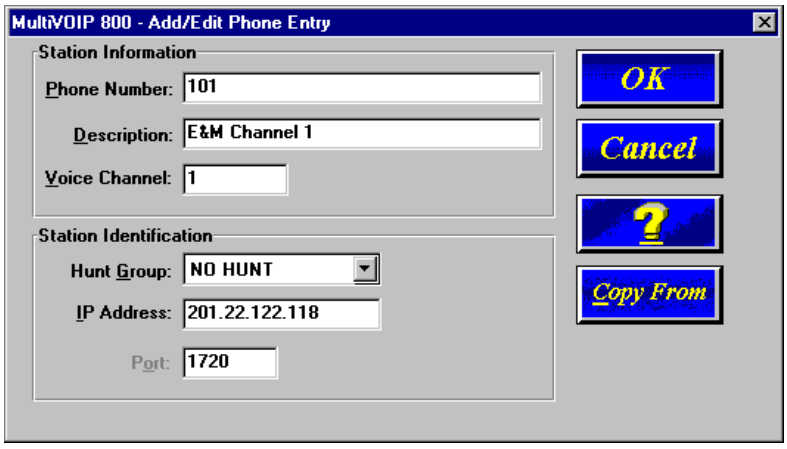


So now when you “come alive”, the Gatekeeper will register you with the above alias address. No other H323 endpoint can use this alias. This is like your own telephone number.

If you select the **Proprietary PhoneBook** option in the **Phone Directory Database** dialog box instead of the **Gatekeeper** option, the **Database Type** group would become active and the **RAS Parameters** group is unavailable.

You can change the typical application to have the Proprietary Phone Book control the call session. When you elect to use the Proprietary PhoneBook, you set up a master-slave relationship. This relationship allows one MultiVOIP to maintain the Phone Directory Database and publish this data

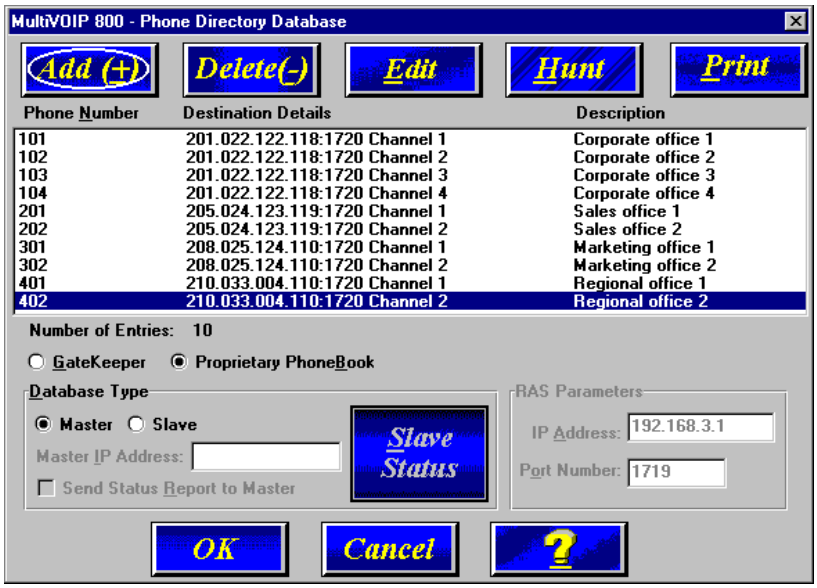
base to all MultiVOIP participants in the network. This proprietary data base enables you to see all the participants in your network and provides you with there phone numbers.



Use the corporate MultiVOIP to set up the database so that the corporate MultiVOIP can call the sales, marketing, or regional offices. To do this, the Phone Directory Database will have four entries for the corporate office, and two entries for the sales office, two for the marketing office and two for the regional office. Extension 101 at the corporate office is tied to voice channel 1, 102 is tied to channel 2, and so on. The Description again ties to the type of interface used on the corporate MultiVOIP (E&M). The Hunt Group in this example is set for No Hunt. But if you wanted to activate a Hunt Group (if an extension on the MultiVOIP is busy and you wanted to look for another extension), you can assign a hunt group to those extensions. For example, if extension 101 is busy, the corporate MultiVOIP would roll over to extension 102.

Again, the IP Address of the corporate MultiVOIP needs to be added and the port number is 1720. This adds phone number 101 of the corporate MultiVOIP to the proprietary data base. Now, to add extension 102 to the proprietary data base, all you have to do is change the Phone Number and Description to support channel 2 of the corporate MultiVOIP. After you have added channel 2, you need to include the two entries for each office.

The proprietary data base would then appear as in the following dialog box and when the office MultiVOIP is turned on, the current data base would be downloaded.



Front Panel Description

The front panel contains three groups of LEDs that provide the status of the Ethernet connection, Voice/Fax channels, and general status of the MultiVOIP. The front panel is shown in Figure 1-3, and a description of each LED follows.

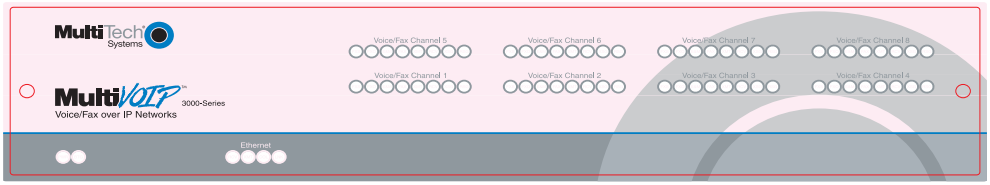


Figure 1-3. Front Panel

ETHERNET

RCV Receive Data indicator blinks when packets are being received from the local area network.

LNK Link indicator lights when the Ethernet link senses voltage from a concentrator or external device.

XMT Transmit Data indicator blinks when packets are being transmitted to the local area network.

COL Collision indicator lights when a collision is detected on the Ethernet link.

VOICE/FAX CHANNEL

FXS Foreign Exchange Station indicator lights when the voice/fax channel is configured for FXS operation.

FXO Foreign Exchange Office indicator lights when the voice/fax channel is configured for FXO operation.

E&M Ear and Mouth indicator lights when the voice/fax channel is configured for E&M operation.

FAX Fax indicator lights when there is fax traffic on the voice/fax channel.

XMT Transmit indicator blinks when voice packets are being transmitted to the local area network.

RCV Receive indicator blinks when voice packets are being received from the local area network.

XSG Transmit Signal indicator lights when the FXS-configured channel is off-hook, the FXO-configured channel is receiving a ring from the Telco, or the M lead is active on the E&M configured channel. That is, the MultiVOIP is receiving a ring from the PBX.

RSG Receive Signal indicator lights when the FXS-configured channel is ringing, the FXO-configured channel has taken the line off-hook, or the E lead is active on the E&M-configured channel.

BTG (BOOT)

The BTG indicator lights when the MultiVOIP is booting or downloading setup.

POWER (PWR)

The Power indicator lights when power is applied to the MultiVOIP.

Back Panel Description

The cable connections for the MultiVOIP are made at the back panel. Connectors include Power, Command Port (RS232), Ethernet (10BASE-T), and Voice/Fax Channels (E&M, FXO and FXS). The cable connectors are shown in Figure 1-4 and defined in the following groups.

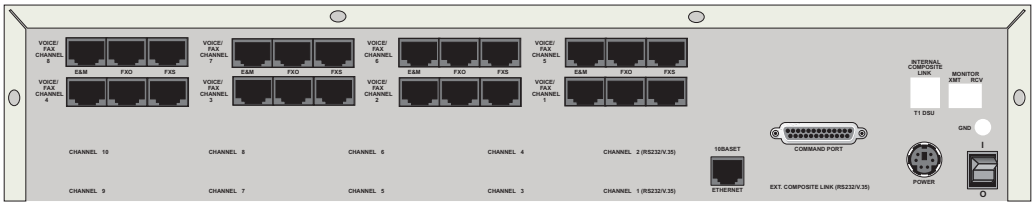


Figure 1-4. Back Panel

Power Connector

The Power connector is used to connect the external power supply to the MultiVOIP. The Power connector is a 7-pin circular DIN connector. A separate power cord is connected to the power supply and the live AC grounded outlet.

Command Port Connector

The Command Port connector is used to configure the MultiVOIP using a PC with a serial port and running Windows® software. The Command Port connector is a DB-25 female connector.

10Base-T (Ethernet) Connector

The Ethernet 10Base-T connector is used to connect the MultiVOIP to a LAN using unshielded twisted cable. This connector is a keyed RJ-45 jack.

Voice/Fax Channel

The Voice/Fax channel connectors include three options per channel: E&M, FXO and FXS.

E&M - This connector is used if you are connecting VOICE/FAX CHANNEL _ to the E&M trunk on a PBX. This connector is an RJ-45 jack.

FXO - This connector is used if you are connecting VOICE/FAX CHANNEL _ to the station side of a PBX. This connector is an RJ-11 jack.

FXS - This connector is used if you are connecting VOICE/FAX CHANNEL _ to a station device, an analog telephone, KTS telephone system, or fax machine. This connector is an RJ-11 jack.

Specifications

- One 1 Meg by 32 byte at 70 nanosecond SIMM is 4 Mb DRAM

  Caution: SIMM speed and size cannot be mixed

- Two Meg of flash memory

Ethernet Port

- Single Ethernet Interface - 10Base-T (twisted pair) keyed RJ-45 connector.

Command Port

- Single 19.2 Kbps asynchronous Command Port with a DB-25 female connector

Voice/Fax Channel

- Two RJ-11 jacks (FXO and FXS)
- One RJ-45 jack (E&M)

Electrical/Physical

- Voltage - 115 VAC (Standard), 240 Volts AC (Optional)
- Frequency - 47 to 63 Hz
- Power Consumption - 18 Watts
- Dimensions - 3.75" high x 17.4" wide x 8" deep

  8.9cm high x 44.2cm wide x 20.3cm deep

- Weight - 7.4 pounds (3.4 kg)

MultiVOIP™

Voice/Fax over IP Networks

Chapter 2 - Installation

MultiTech® Systems

Installing Your MultiVOIP

The basic steps of installing your MultiVOIP 400 network involve unpacking the units, connecting the cables, and configuring the units using management software (MultiVOIP Configuration). This process results in a fully functional Voice Over IP network. A general description is provided below and detailed instructions are provided in Chapter 3, Software Loading and Configuration.

Installing and Configuring Your MultiVOIP

The VOIP administrator must first install the MultiVOIP 400 software and then configure each MultiVOIP 400 for its specific function. During the configuration process, it is important to note that the Phone Directory Database is configured differently depending on whether or not you have Gatekeeper support on your VOIP network.

If your VOIP network supports Gatekeeper software, you must preregister or remotely register all H.323 endpoints with the Gatekeeper. The procedure for doing this is explained in the “Registering with a Gatekeeper Phone Directory” section.

If your VOIP network does not have Gatekeeper software or the Gatekeeper software is not enabled, then you must build a proprietary phonebook with a master MultiVOIP and slave MultiVOIPs. The master unit includes the assignment of a unique LAN IP address, subnet mask, and Gateway IP address; as well as the selection of appropriate channel interface type for each of the Voice/Fax channels. Once all connections have been made, the VOIP administrator configures the unit and builds the Phone Directory Database that will reside with the master unit.

Once configuration of the master MultiVOIP has been completed, the administrator can configure the MultiVOIPs designated as slave units. Again, unique LAN IP addresses, subnet masks, and Gateway IP addresses are assigned, and each Voice/Fax channel is configured for the appropriate channel interface type. When this is done, the Phone Directory Database option is set to **Slave**, and the IP address of the master MultiVOIP is entered. Once all slave units are configured, the process moves on to the “Deploying the VOIP Network” section.

Deploying the VOIP Network

The final phase of the installation is deployment of the network. When the remote MultiVOIPs are sent to their remote sites, the remote site administrators need only to connect the units to their LAN and telephone equipment. A full Phone Directory Database (supplied by the master MultiVOIP Proprietary Phonebook will be loaded into their units within minutes of being connected and powered up. For remote VOIPs that were configured with the **Gatekeeper** option enabled, each MultiVOIP will be remotely registered with the Gatekeeper. The Gatekeeper phonebook directory is NOT downloaded to the remote units).

Safety Warning Telecom

1. Never install telephone wiring during a lighting storm.
2. Never install telephone jacks in wet locations unless the jack is specifically designed for wet locations.
3. This product is to be used with UL and cUL listed computers.
4. Never touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.
5. Use caution when installing or modifying telephone lines.
6. Avoid using a telephone (other than a cordless type) during an electrical storm. There may be a remote risk of electrical shock from lighting.
7. Do not use the telephone to report a gas leak in the vicinity of the leak.
8. To reduce the risk of fire, use only No. 26 AWG or larger telecommunications line cord.

Unpacking Your MultiVOIP

Remove all items from the box.

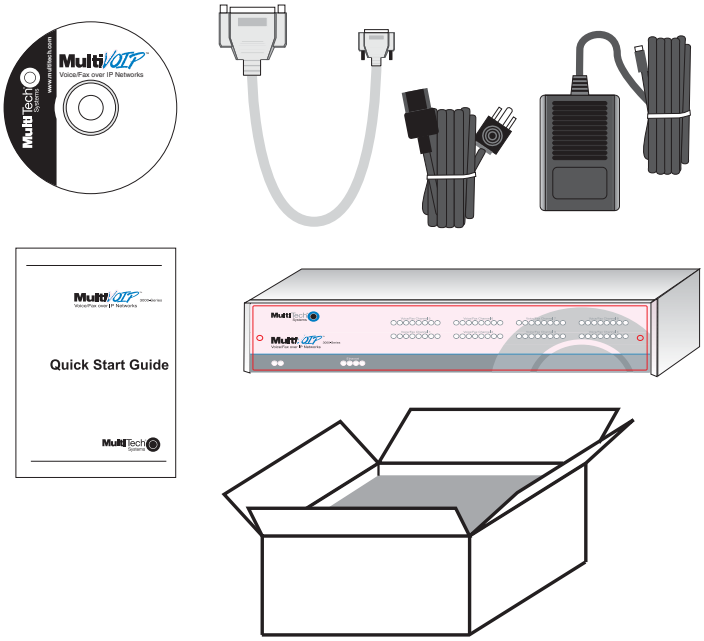


Figure 2-1. Unpacking

Safety Warnings

Caution: Danger of explosion if battery is incorrectly replaced.

A lithium battery on the voice/fax channel board provides backup power for the time keeping capability. The battery has an estimated life expectancy of ten years.

When the battery starts to weaken, the date and time may be incorrect. If the battery fails, the board must be sent back to Multi-Tech Systems for battery replacement.

The E&M, FXS, and Ethernet ports are not designed to be connected to a Public Telecommunication Network.

Cabling Procedure

Cabling involves connecting the master MultiVOIP to your LAN and telephone equipment.

1. If you are connecting any Voice/Fax Channel to an E&M trunk other than type 2, perform the E&M Jumper Block Positioning procedure on the next page before connecting power to the unit.
2. Connect one end of the power supply to a live AC outlet and connect the other end to the MultiVOIP as shown in Figure 2-2. The power connector is a 7-pin circular DIN connector.

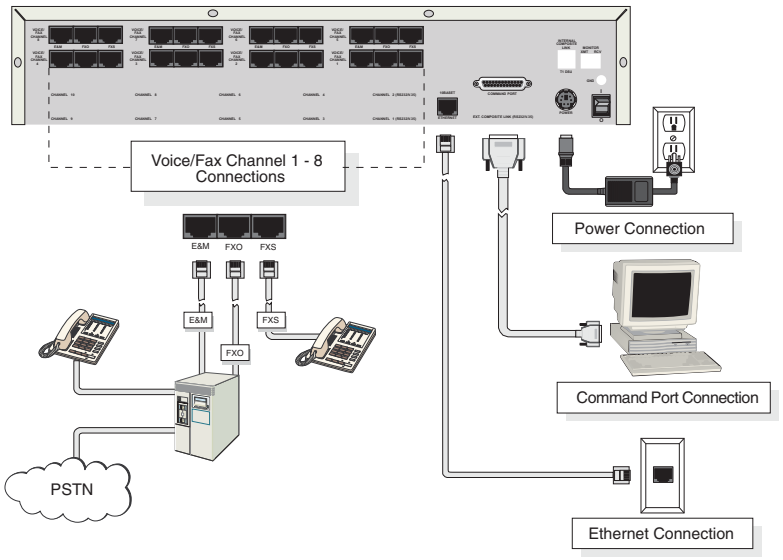


Figure 2-2. Cable Connections

3. Connect the MultiVOIP to a PC by using a DB-25 (male) to DB-9 (female) cable. Plug the DB-25 end of the cable into the Command port of the MultiVOIP and the other end into the PC serial port. See Figure 2-2.
4. Connect a network cable to the ETHERNET 10BASET connector on the back of the MultiVOIP. Connect the other end of the cable to your network.
5. If you are connecting a station device such as an analog telephone, a fax machine, or a Key Telephone System (KTS) to your MultiVOIP, connect one end of an RJ-11 phone cord to the **Voice/Fax Channel 1 FXS** connector on the back of the MultiVOIP and the other end to the station device.

   If you are connecting the station side of a telephone switch (PBX) to your MultiVOIP, connect one end of an RJ-11 phone cord to the **Voice/Fax Channel 1 FXO** connector on the back of the MultiVOIP and the other end to the phone jack.

   If you are connecting an E&M trunk from a telephone switch to your MultiVOIP, connect one end of an RJ-45 phone cord to the **Voice/Fax Channel 1 E&M** connector on the back of the MultiVOIP and the other end to the trunk. Refer to Appendix B for E&M cabling pinout.

   If you are connecting to an E&M trunk, verify that the E&M trunk jumper is in the correct position for the E&M type trunk. The default E&M jumper position is E&M type 2. To change the E&M jumper position, perform the E&M jumper block positioning procedure.
6. Repeat the above step to connect the remaining telephone equipment to each Voice/Fax Channel on your MultiVOIP.
7. Turn on power to the MultiVOIP by placing the ON/OFF switch on the back panel to the ON position. Wait for the **BTG** LED on the MultiVOIP to go off before proceeding. This may take a few minutes.

   Proceed to Chapter 3 to load the MultiVOIP software.

E&M Jumper Block Positioning Procedure

A jumper block exists for each voice/fax channel. The jumper block is to the right of each set of channel jacks. The jumper block contains 8 pairs of pins. The jumper plug fits over three pairs of pins on the jumper block. The E&M type number is labeled on the PC board. The jumper plug needs to be centered on the E&M type number. Perform the following procedure to change E&M jumper position.

1. Ensure that power is removed from the MultiVOIP
2. Remove the front panel by loosening the two Phillips quarter turn screws.
3. Remove six Phillips screws and slide the top cover back off the chassis to expose the rear panel.
4. To change a jumper position, lift the jumper plug up off the jumper block and move to the new position, ensuring that the center jumper is centered on the E&M type number.

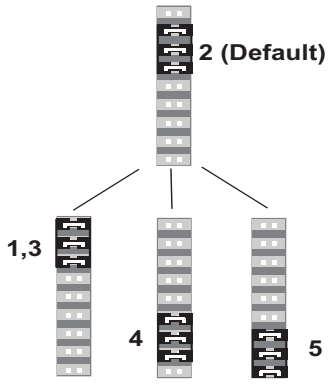
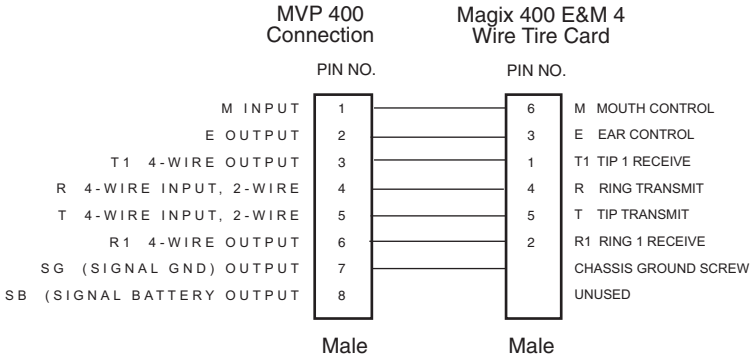


Figure 2-3. E&M Jumper Block

5. Change the jumper position for each voice/fax channel that is connecting to an E&M trunk that is not a type 2.
6. Slide the top cover back on to the chassis and replace the six Philips screws.
7. Replace the front panel and secure it by tightening the two Phillips quarter turn screws.
8. If you are using a Magix 400 E&M Tie Card, connect the ground pin to the chassis ground screw as shown.



9. Return to the Cabling Procedure section.

MultiVOIP™

Voice/Fax over IP Networks

Chapter 3 - Software Loading and Configuration

MultiTech® Systems

Loading Your MultiVOIP Software

The following software loading procedures do not provide every screen or option in the process. It is assumed that a technical person with a thorough knowledge of Windows and the software loading process is doing the installation. Once you have loaded the software, you will be instructed on how to configure your MultiVOIP 400, and finally, on how to deploy your MultiVOIP 400. Additional information on the MultiVOIP 400 software is provided in the Help.

Note: The phonebook directory configuration process is different depending on whether or not you have an enabled H.323 Gatekeeper resident in your network. The “Configuring Your MultiVOIP 400” section will explain these differences.

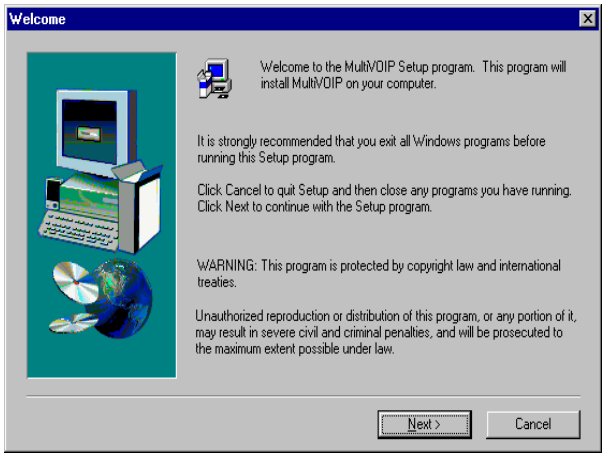
CAUTION: If you are installing a MultiVOIP 400 behind a Firewall, the Firewall must support H.323. Refer to your Firewall user documentation to enable H.323 support.

1. Verify that your MultiVOIP 400 has been properly cabled and that it is powered on.
2. Insert the MultiVOIP CD into your CD-ROM drive. The CD should start automatically. It may take 10 to 20 seconds for the MultiVOIP 400 Installation CD window to display.

If the MultiVOIP MVP400 window does not appear automatically, click **My Computer**, then right-click the **CD-ROM** drive icon, click **Open**, and then click the **Autorun** icon.

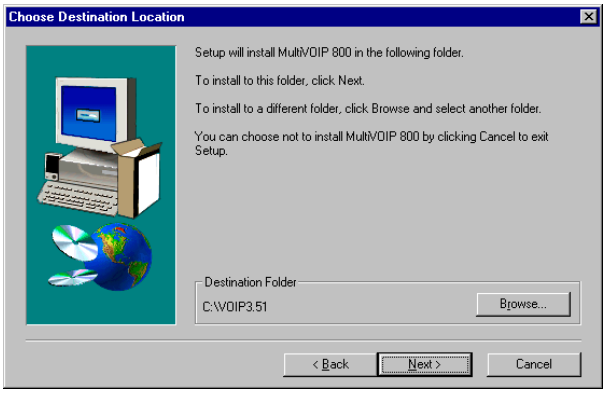
3. When the MultiVOIP MVP400 window displays, click the **Install Software** icon.
4. The Select Software dialog box is displayed. Select the **H.323 Compatible** option and click **OK**.

5. The **MultiVOIP 800 Setup** welcome dialog box displays.


Welcome
Welcome to the MultiVOIP Setup program. This program will install MultiVOIP on your computer.
It is strongly recommended that you exit all Windows programs before running this Setup program.
Click Cancel to quit Setup and then close any programs you have running. Click Next to continue with the Setup program.
WARNING: This program is protected by copyright law and international treaties.
Unauthorized reproduction or distribution of this program, or any portion of it, may result in severe civil and criminal penalties, and will be prosecuted to the maximum extent possible under law.
Next >
Cancel

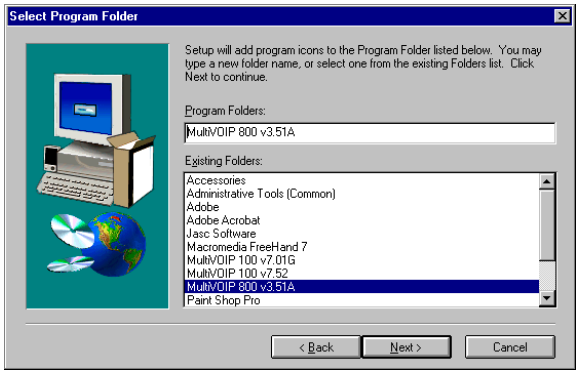
Press **Enter** or click **Next** to continue.

6. The **Choose Destination Location** dialog box displays. Follow the instructions.


Choose Destination Location
Setup will install MultiVOIP 800 in the following folder.
To install to this folder, click Next.
To install to a different folder, click Browse and select another folder.
You can choose not to install MultiVOIP 800 by clicking Cancel to exit Setup.
Destination Folder
C:\VOIP3.51
Browse...
< Back
Next >
Cancel

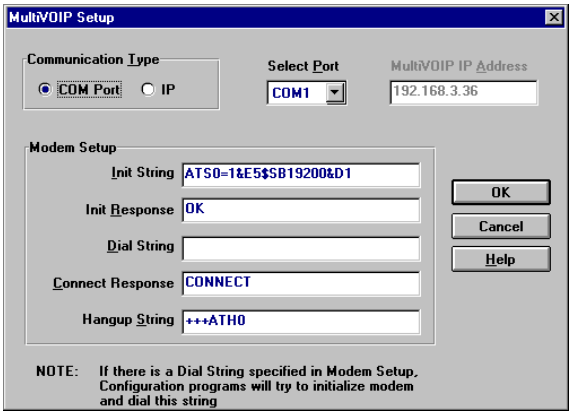
You can either choose the Destination Location of your MultiVOIP 400 software or select the default destination by clicking **Next**. Click **Browse** if you want to select a different destination folder for the MultiVOIP 400 software.

7. The **Select Program Folder** dialog box enables you to choose where you want the program file to be located.


Select Program Folder
Setup will add program icons to the Program Folder listed below. You may type a new folder name, or select one from the existing Folders list. Click Next to continue.
Program Folders:
MultiVOIP 800 v3.51A
Existing Folders:
Accessories
Administrative Tools (Common)
Adobe
Adobe Acrobat
Jasc Software
Macromedia FreeHand 7
MultiVOIP 100 v7.01G
MultiVOIP 100 v7.52
MultiVOIP 800 v3.51A
Paint Shop Pro
< Back
Next >
Cancel

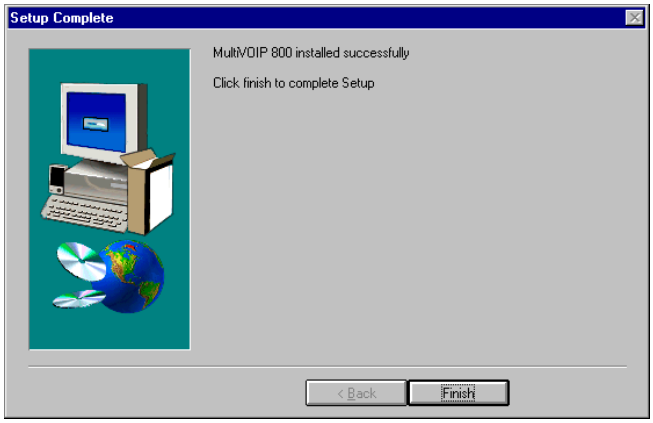
Accept the default program folder name or enter another name in the **Program Folder** box. Click **Next** to continue.

8. The **MultiVOIP 800 Setup** dialog box displays. This dialog box enables you to select the COM port of your PC that is connected to the Command port of the MultiVOIP 400. From the **Select Port** list, select the COM port of your PC.



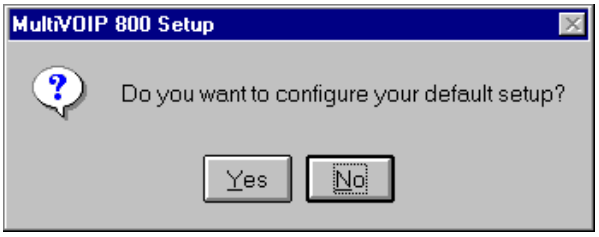
Click **OK** to continue.

9. The **Setup Complete** dialog displays.



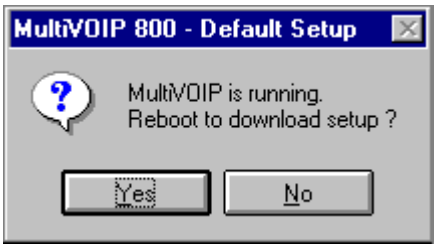
Click **Finish** to continue.

10. The following message displays:



Click **Yes** to continue.

11. The following message displays:



Click **Yes** to continue.

Configuring Your MultiVOIP 400

The following steps provide instructions for configuring your MultiVOIP 400. The configuration sequence includes IP Protocol default setup, Channel setup, and Phone Directory Database setup. The Phone Directory Database setup is configured differently depending on whether or not the Gatekeeper function is available and enabled on the **Phone Directory Database** dialog box (see Step 26).

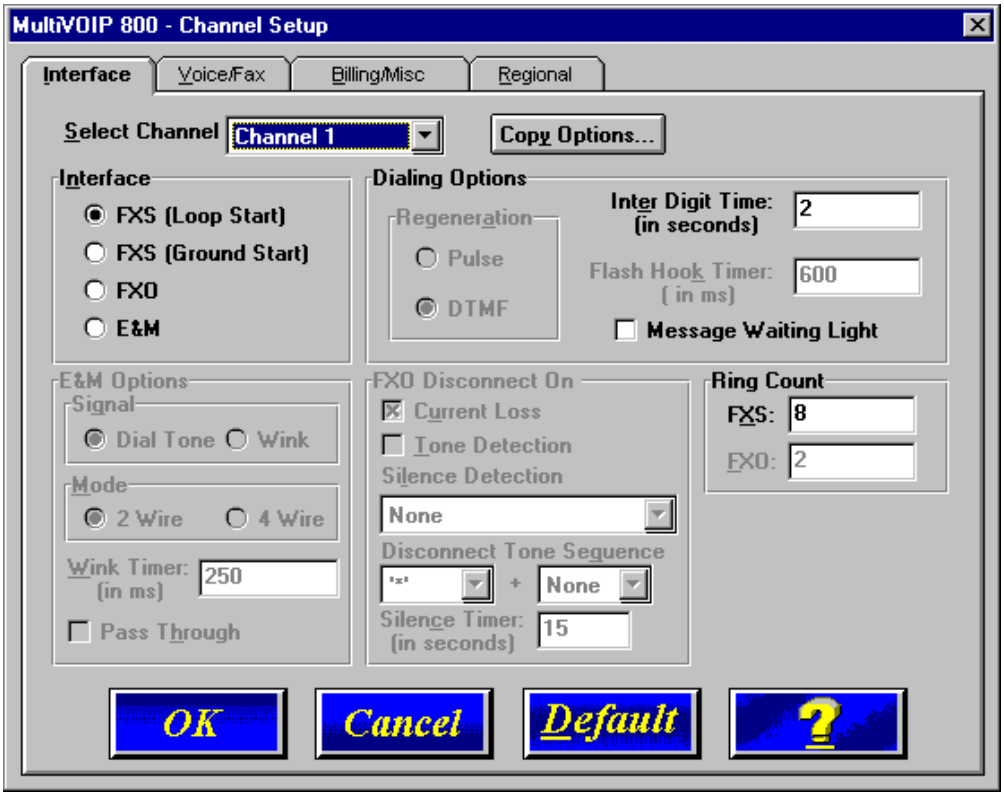
12. The **IP Protocol Default Setup** dialog box displays.

The default **Frame Type** is **TYPE_II**. If this does not match your IP network, change the Frame Type by selecting **SNAP** from the **Frame Type** list. The available Frame Type choices are **TYPE_II** and **SNAP**.

13. In the **Ethernet** group, enter the **IP Address**, **Subnet Mask**, and **Gateway Address** unique to your IP LAN. The IP address is the unique LAN IP address that is assigned to the MultiVOIP 400, and the Gateway address is the IP address of the device connecting your MultiVOIP 400 to the Internet.

    Click **OK** when you are finished.

14. The **Channel Setup** dialog box displays. The four tabs in this dialog box define the channel interface, voice/fax parameters, billing, miscellaneous, and regional telephone parameters for each channel.



Configure each channel for the type of interface you are using. **Channel 1** is selected by default in the **Select Channel** list. To change the channel number, select the channel you want to configure from the list.

Note: Feature options are enabled or unavailable depending on the selected interface type. The one option available for all interface types is the **Inter Digit Time** option. This option defines the maximum amount of time that the unit will wait before mapping the dialed digits to an entry in the Phone Directory Database. If too much time elapses between digits, and the wrong numbers are mapped, you will hear a rapid busy signal. If this happens, hang up and dial again. The default setting is **2** seconds.

15. The **Interface** group defaults to **FXS (Loop Start)**. Select the interface option that corresponds to the interface type being connected to the Voice/Fax Channel 1 jack on the back panel of the MultiVOIP 400.

    FXS (Loop Start): If a station device such as an analog telephone, fax machine, or KTS (Key Telephone System) is connected to the Voice/Fax connector on the back of the unit, FXS (Loop Start) will likely be the correct Interface.

    FXS (Ground Start): If the station device uses ground start, then choose the FXS (Ground Start) option. Refer to the device's user documentation.

    For both FXS Loop Start and FXS Ground Start, the **Ring Count FXS** window enables you to set the maximum number of rings output on the FXS interface before hanging up and releasing the line to another call. The default setting is **8** rings.

    Note: Zero (0) means no rings. The caller hears a busy tone.

    FXO: If you are using an analog extension from your PBX, then choose the FXO option. Check with your in-house phone personnel to verify the connection type.

    If FXO is selected, the **Dialing Options Regeneration, Flash Hook Timer,** and **Ring Count** groups are enabled. Check with your local in-house phone personnel to verify whether your local PBX dial signaling is **Pulse** or tone (**DTMF**). Then, set the Regeneration option accordingly. The **Flash Hook Timer** allows you to enter the time, in milliseconds, for the duration of the flash hook signals output on the FXO interface. The default setting is **600** milliseconds. The **Ring Count FXO** window allows you to set the number of rings received on the FXO interface before the MultiVOIP 400 answers the incoming call. The default setting is **2** rings.

    Note: Zero (0) means that the MultiVOIP 400 never answers.

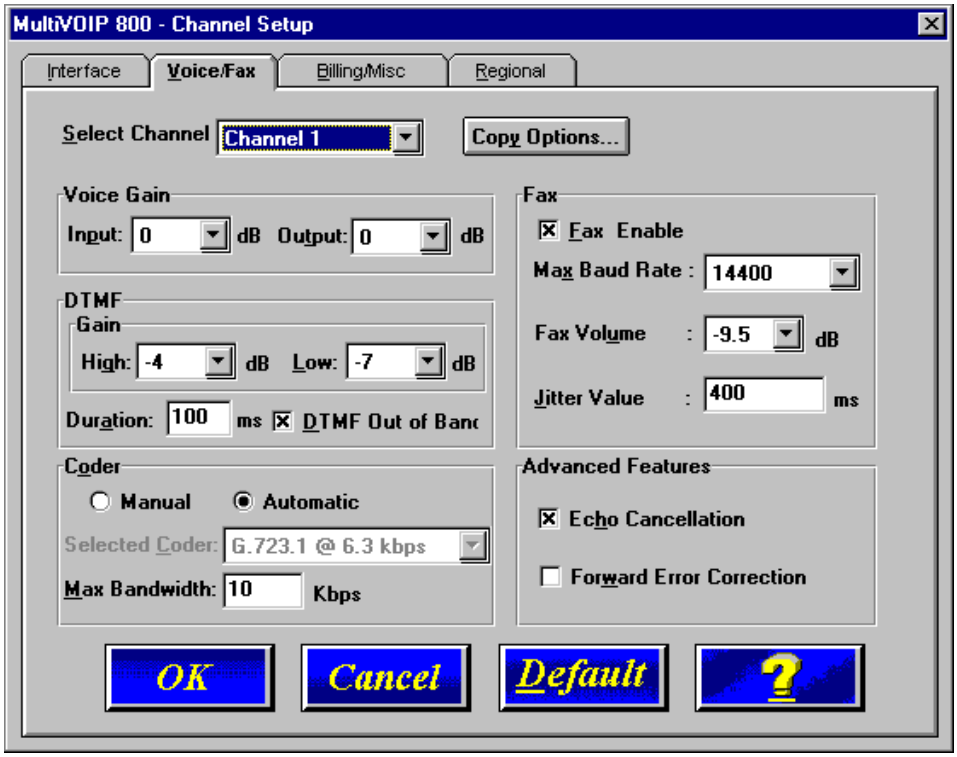
    For FXO-to-FXO communications, you can enable a specific type of FXO Disconnect: **Current Loss**, **Tone Detection**, or **Silence Detection**. Check with your in-house phone personnel to verify the preferred type of disconnect to use. Enabling **Tone Detection** activates the Disconnect Tone Sequence options. For **Disconnect Tone Sequence**, you can select either one or two tones that will disconnect the line. The person hanging up a call must then hit the key or keys that will produce those tones. For **Silence Detection**, select **One Way** or **Two Way**, then set the timer for the number of seconds of silence before disconnect. The default value of **15** seconds may be shorter than desired for your application.

    E&M: If you are connecting to an analog E&M trunk on your PBX, then choose the E&M interface option to enable the E&M Options group. Check with your in-house phone personnel to determine if the signaling is **Dial Tone** or **Wink** and if the connection is 2-wire or 4-wire. If Wink signaling is used, then the **Wink Timer** is enabled with a default of **250** milliseconds. The range of the Wink Timer is from 100 to 350 milliseconds. Consult with your local in-house phone personnel for this timer setting.

    Note: After configuring a given channel, you can copy that channel configuration by clicking **Copy**. Everything on the **Interface** tab will be copied to the other channel.

16. Repeat the above step to configure the interface type for each voice/fax channel.

17. The **Voice/Fax** tab displays the parameters for the voice gain, DTMF (Dual Tone Multi-Frequency) gain, voice coder, faxing, and advanced features such as Silence Compression, Echo Cancellation, and Forward Error Correction.



18. You can set the input and output voice gain so that the volume can be increased or decreased. Input gain modifies the level of the audio coming in to the voice channel before it is sent over the Internet to the remote MultiVOIP 400. Output gain modifies the level of the audio being output to the device attached to the voice channel. Make your selections from the Input and Output lists in the **Voice Gain** group. The valid range is +31dB to –31dB with a recommended default value of **0**.

    You can set up the DTMF gain (output level in decibels) for the higher and lower frequency groups of the DTMF tone pair. Make your selections from the **High** and **Low** lists in the **Voice Gain** group. When the **DTMF Out of Band** check box is selected, the unit reproduces the DTMF tones instead of passing them through.

    Note: The DTMF gain should be changed only with the assistance of Multi-Tech Technical Support.

19. To change the voice coder, select the channel from the **Select Channel** list, then click **Manual** in the Coder group. Select the new voice coder entry from the **Selected Coder** list.

    If you changed the voice coder, ensure that the same voice coder is used on the voice/fax channel you are calling. Otherwise, you will always get a busy signal.

    Note: If you allow the Coder to be selected automatically, then you need to enter the **Max Bandwidth**. Check with your network administrator to determine how much bandwidth is available.

20. The **Fax** group enables you to send/receive faxes on the selected voice/fax channel. You can set the maximum baud rate for faxes, the fax volume, and enter the jitter value in milliseconds.

    When receiving fax packets from a remote MultiVOIP 400, it is possible for individual packets to be delayed or received out of order due to traffic conditions on the network. To compensate for this effect, the MultiVOIP 400 uses a Jitter Buffer. The **Jitter Value** enables the MultiVOIP 400 to wait a user-definable period of time, in milliseconds, for delayed or out of order fax packets. The range of allowable Jitter Values is 0 to 400 milliseconds, with a default of **400** milliseconds.

If you do not plan to send or receive faxes on a given voice/fax channel, you can disable faxes in the **Fax** group.

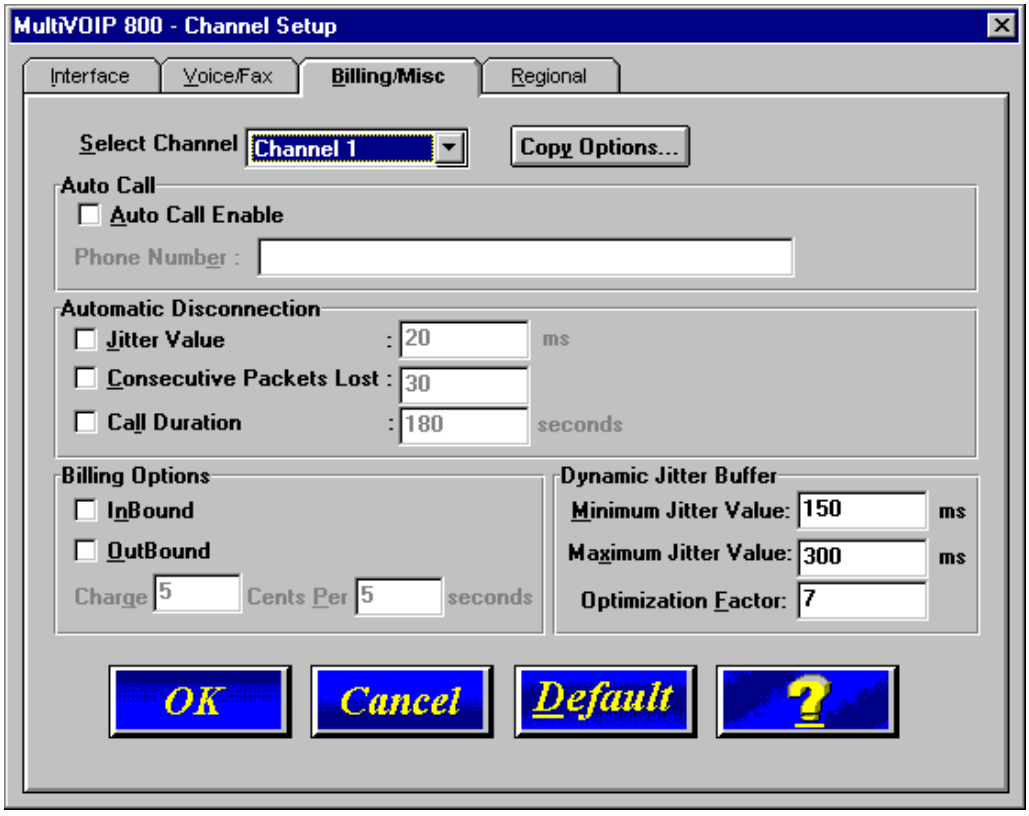
21. You can enable the voice/fax advanced features by selecting **Echo Cancellation** or **Forward Error Correction** check boxes.

    The **Echo Cancellation** check box defines whether echo cancellation is enabled for this voice channel. If echo cancellation is enabled, the MultiVOIP 400 will remove echo which improves the quality of sound.

    The **Forward Error Correction (FEC)** check box defines whether forward error correction is enabled for this voice channel. The FEC feature allows some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel.

    Note: After configuring a given channel, you can copy that channel configuration by clicking **Copy**. Everything on the **Voice/Fax** tab will be copied to the other channel.

22. The **Billing/Misc** tab displays the parameters for auto call, automatic disconnection, billing options, and dynamic jitter buffer.



If you want to dedicate a local voice/fax channel to a remote voice/fax channel (so you will not have to dial the remote channel), select the **Auto Call Enable** check box in the **Auto Call** group. Then enter the phone number of the remote MultiVOIP 400 in the **Phone Number** box.

23. The **Automatic Disconnection** group provides three options to be used singly or in combination.

    The **Jitter Value** defines the average inter-arrival packet deviation in milliseconds before the call is automatically disconnected. Jitter is the inter-arrival packet deviation (phase shift of digital pulses) over the transmission medium that causes voice breakup which can be particularly disruptive to voice communications. The default setting is 20 milliseconds. A higher value means that the voice transmission will be more accepting of jitter. A lower value will be less tolerant of jitter.

Consecutive Packets Lost defines the number of consecutive packets that are lost after which the call is automatically disconnected. The default setting is **30**.

Call Duration defines the maximum length of time in seconds that a call remains connected before being automatically disconnected. The default setting is **180** seconds. A call limit of three minutes may be too short for most configurations. Therefore, you may want to increase this default value.

24. You can set billing options for inbound and/or outbound calls by selecting them in the **Billing Options** group and then entering the charge in cents per number of seconds.

25. A minimum and maximum set of values can be set for **Dynamic Jitter Buffer**. When receiving voice packets from a remote MultiVOIP 400, it is possible to experience varying delays between packets due to traffic conditions on the network. This is called Jitter. To compensate for this effect, the MultiVOIP 400 uses a Dynamic Jitter Buffer. The Jitter Buffer allows the MultiVOIP 400 to wait for delayed voice packets by automatically adjusting the length of the Jitter Buffer between configurable minimum and maximum values. An **Optimization Factor** adjustment controls how quickly the length of the Jitter Buffer is increased when jitter increases on the network. The length of the jitter buffer directly effects the voice delay between MultiVOIP 400 gateways.

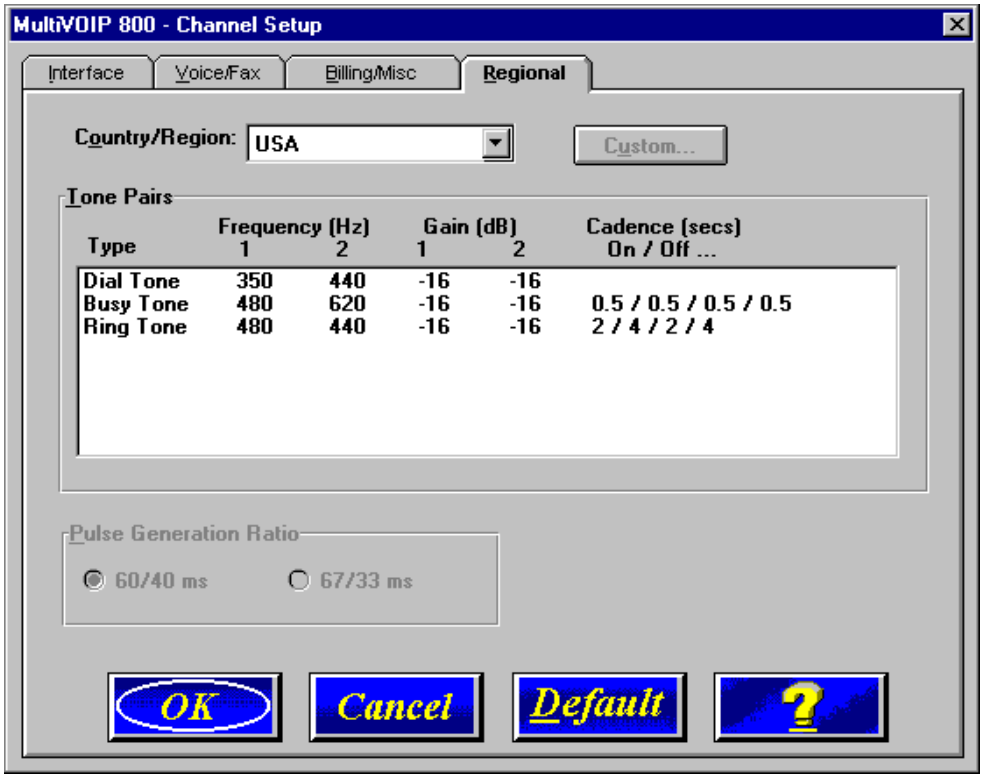
    The Minimum Jitter Value default setting is **150** milliseconds, the Maximum Jitter Value default setting is **300** milliseconds, and the Optimization Factor default setting is **7**.

    Note: After configuring a given channel, you can copy that channel configuration by clicking **Copy Options**. Everything on the **Billing/Misc** tab will be copied to the other channel.

    If your country/region is not USA, the default , click the **Regional** tab and proceed to step 26. Otherwise, proceed to step 27 to begin building your phone directory database.

26. To change the Tone Pairs on the **Regional** tab, select your country or region from the **Country/ Region** list.

    Note: If your country or region is not listed, click **Custom** to define it.



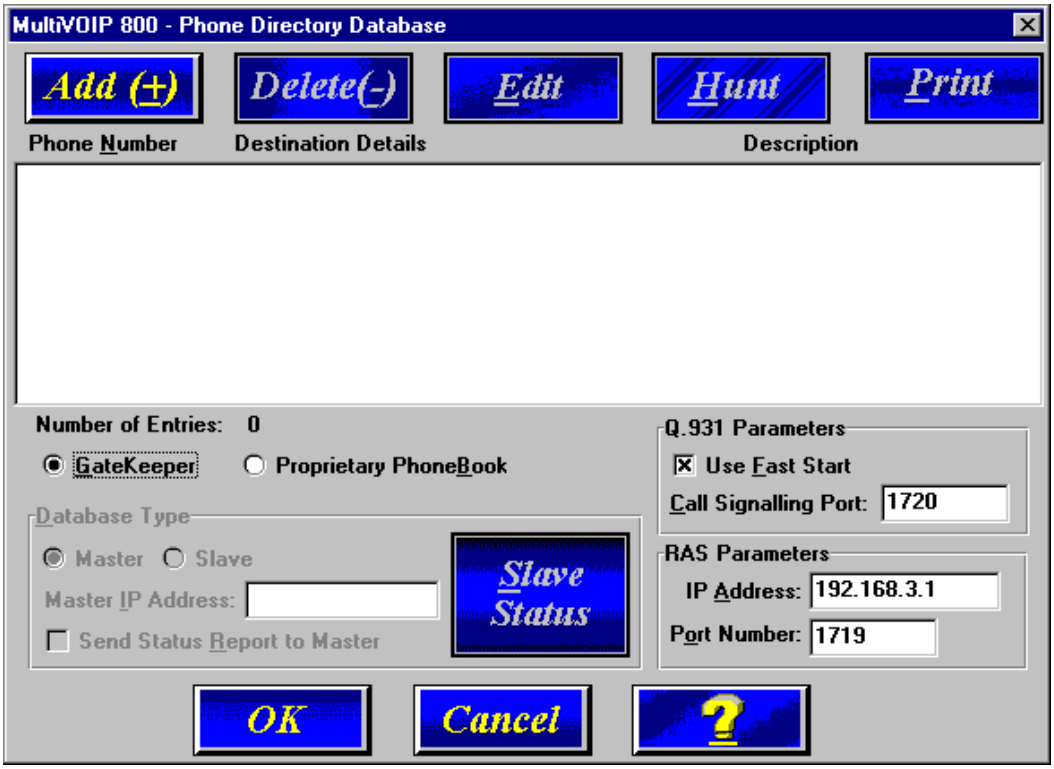
The **Tone Pairs** group enables you to select and modify the parameters. Click **OK** when finished.

27. The **Phone Directory Database** dialog box displays. This dialog box enables you to select either the GateKeeper or Proprietary Phone Book. Once you have selected the type of Phone Book database, you can proceed to “Registering with a Gatekeeper Phone Directory” in the following section. Or, if you are building a proprietary phone book, proceed to the “Building a Proprietary Phonebook Directory” section later in this manual.

Registering with a Gatekeeper Phone Directory

This section describes how to register H.323 endpoints with the GateKeeper. The H.323 GateKeeper function resides at a PC acting as the central point for all calls within its zone and provides call control services to registered endpoints. The GateKeeper performs two important call control functions: address translation from LAN aliases to IP addresses, and bandwidth management where the network manager has specified a threshold for the number of simultaneous conferences on the LAN.

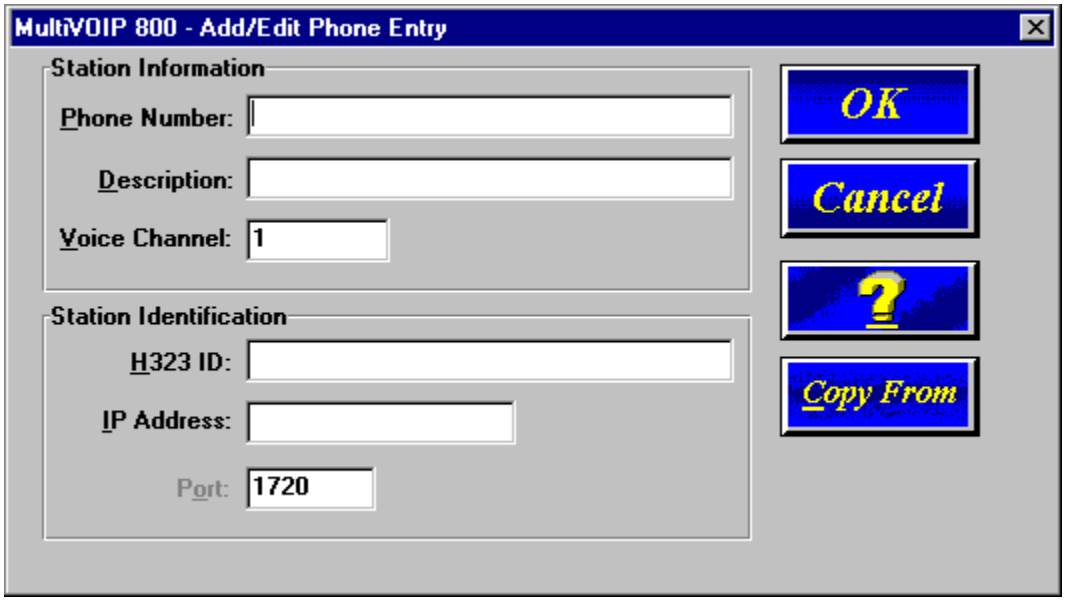
In a GateKeeper environment, you will be selecting the **GateKeeper** option, entering an IP address for the GateKeeper, and accepting the default port number. If the GateKeeper network is servicing Fast Start, accept the defaults in the **Q.931 Parameters** group. If this network zone is primarily non-Fast-Start supported, you will clear the **Use Fast Start** check box.



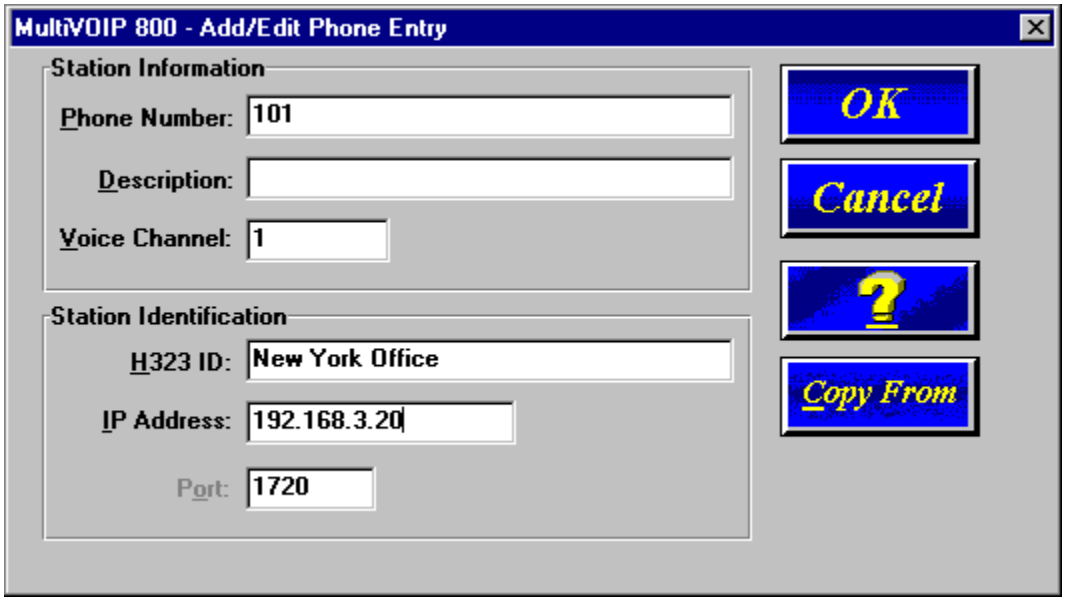
1. Select the **Gatekeeper** option.
2. The **Fast Start** option is selected by default. If the GateKeeper network employs Fast Start, leave this selected. You may have to verify this with the GateKeeper administrator.
3. Enter the GateKeeper IP Address in the **IP Address** box of the **RAS Parameters** group.
4. Accept the default **Port Number 1719**.

   CAUTION: The default setting for the GateKeeper Port Number is 1719. This can be changed to a different value by the GateKeeper administrator. If you decide to change the default Port Number, you must use the same number on the GateKeeper and all other H.323 endpoints.

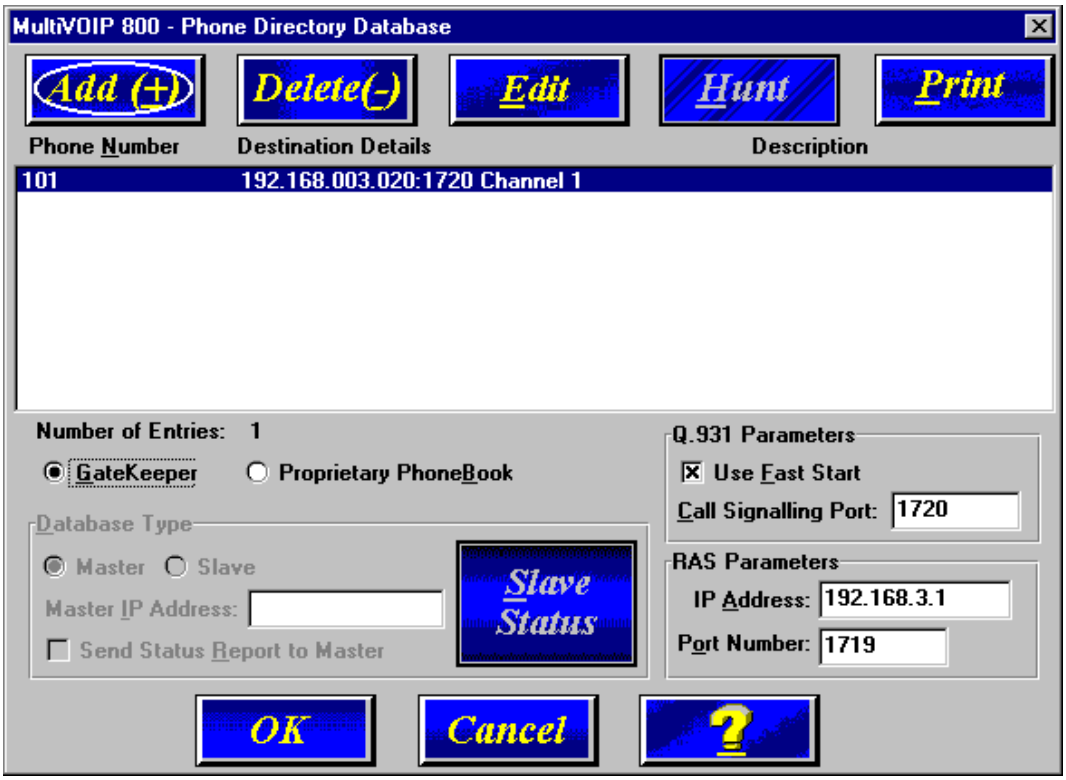
5. Click **Add** to begin building your phone directory database. The **Add/Edit Phone Entry** dialog box displays.



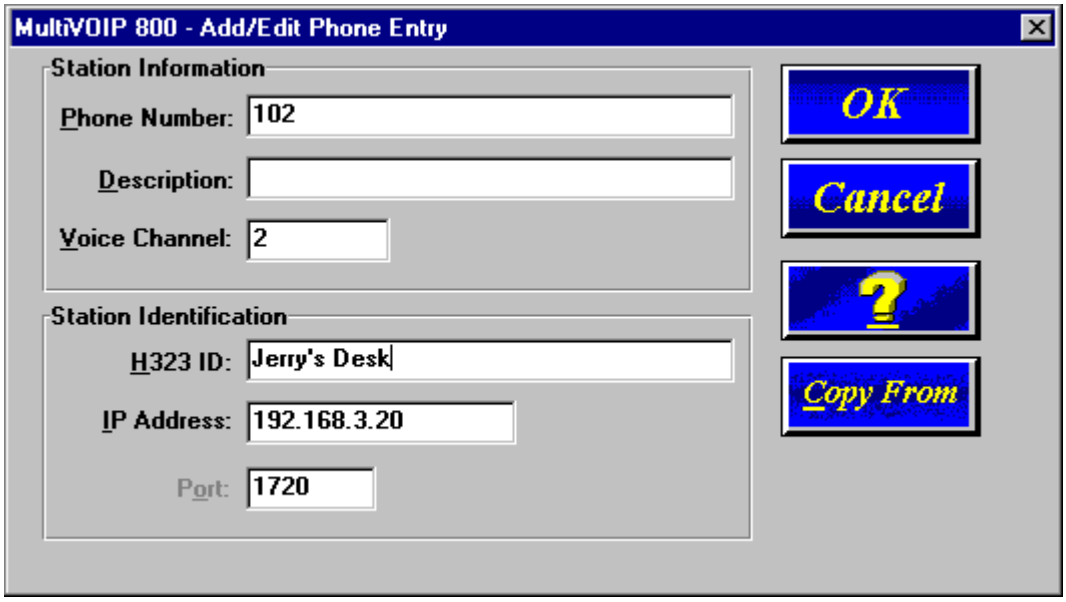
6. Enter the unique phone number of the local device in the **Phone Number** box. For example, 101.
7. Leave the **Description** box blank.
8. Enter the **Voice Channel** number that corresponds to the phone number entered.
9. Enter a description in the **H.323 ID** box to identify the phone number. For example, “New York Office 1”.
10. Enter the Port IP Address of the MultiVOIP you are currently configuring in the **IP Address** box.



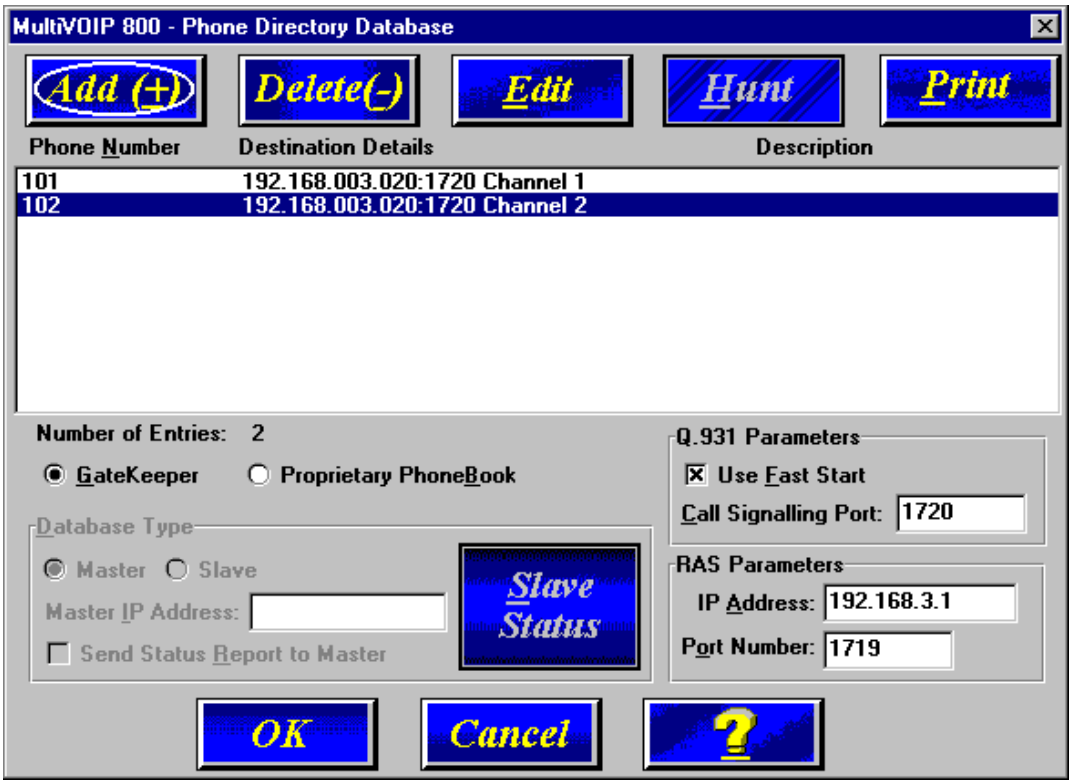
11. Click **OK** when you are finished. The **Phone Directory Database** dialog box displays your first entry.



12. Click **Add** to enter the next phone listing. The **Add/Edit Phone Entry** dialog box displays.
13. Enter the second unique phone number of the local device in the **Phone Number** box. For example, 102.
14. Leave the **Description** box blank.
15. Enter the **Voice Channel** number corresponding to the phone number entered.
16. Enter a description in the **H.323 ID** box to identify the phone number. For this example, you could enter a description such as “Jerry’s Desk”.
17. Enter the Port IP Address of the MultiVOIP you are currently configuring in the **IP Address** box.



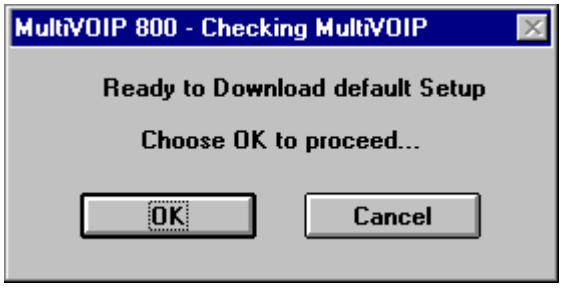
18. Click **OK** when you are finished. The **Phone Directory Database** dialog box displays with your second entry in the window.



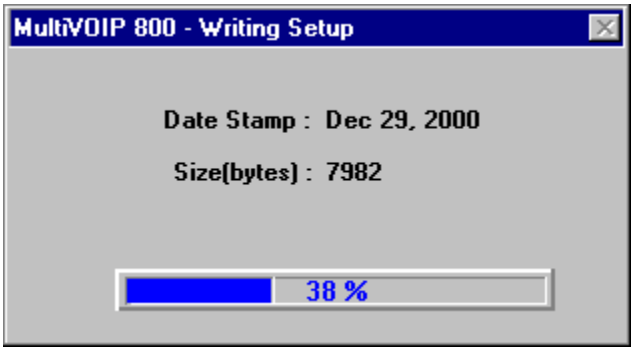
19. Repeat this process for all channels and then click **OK** in the **Phone Directory Database** dialog box.

    Note: When entering multiple channels, you can also click **Copy From** in the **Add/Edit Phone Entry**, enter the phone number you want to copy, click **OK** in the **COPY FROM** dialog box. Change the information as needed, and click **OK** in the **Add/Edit Phone Entry** dialog box.

20. The following dialog box displays. Click **OK** to download setup.



21. Once the setup program receives a response from the MultiVOIP 400, the **Writing Setup** dialog box displays, indicating that the setup configuration is being written to the MultiVOIP 400.



22. After the setup has been written to the MultiVOIP 400, the unit is rebooted.

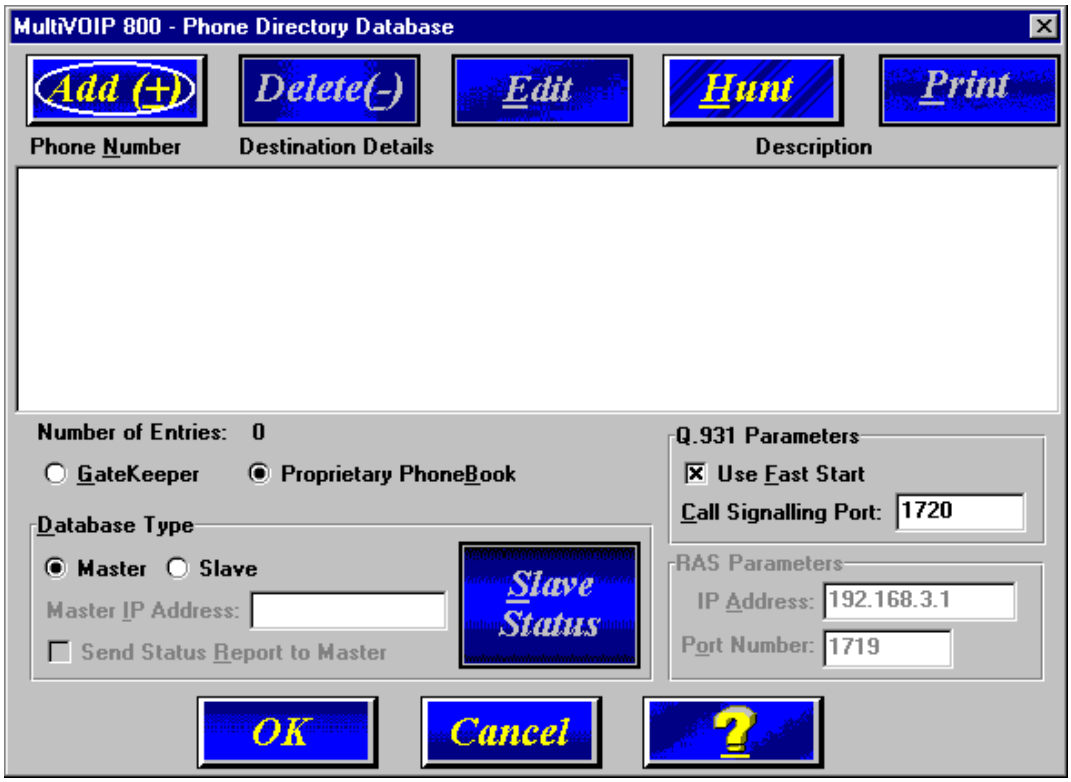
23. Verify that the **BTG** LED on the MultiVOIP 400 is off after the download is complete. This may take several minutes as the MultiVOIP 400 reboots.

24. You are returned to the MultiVOIP MVP400 screen from which you can load the Acrobat Reader to your PC. This allows you to view and/or print the User Guide by clicking the **Install Manuals** or **View Manuals** icon.

Building a Proprietary Phonebook Directory

To build your proprietary MultiVOIP 400 Phone Directory (in an H.323 environment without the **GateKeeper** option enabled), select the **Proprietary Phonebook** option and then configure the "master" MultiVOIP 400 and then add the "slave" MultiVOIP 400s (or other H.323 endpoints). Configuring the "slave" MultiVOIP 400 is discussed later.

The first MultiVOIP 400 to be configured is designated the "master" and contains the proprietary phonebook database. All subsequent MultiVOIP 400s added to the proprietary phonebook database are designated "slaves". The master database contains the phone numbers of all H.323 endpoints available for communication on an IP network. This database is downloaded to each slave MultiVOIP 400 as it comes online.



To configure the "master" MultiVOIP 400, verify that the **Proprietary Phonebook** and **Master** options are selected. The **Master IP Address**, **Send Status Report to Master**, and **RAS Parameters** group will be unavailable.

Note: Click **Slave Status** to display the **Slave VOIP Status** dialog box used for viewing phone number, IP address, status, and description of slave units. See the "Configuring Your Slave MultiVOIP 400s" section for details.

1. Click **Add** to begin building your phone directory database. The **Add/Edit Phone Entry** dialog box displays.

2. Enter the unique phone number of the local device in the **Phone Number** box (for example, 101) and indicate that the local device is connected to Channel 1 in the **Voice Channel** box.

3. The **Description** box is optional, but can be useful in associating the channel to the extension. Enter a description of your local phone number, if desired. This description helps identify the phone number you entered in the previous step. For example, the master MultiVOIP 400 normally resides at the entity's main office. In this example, you could enter a description such as "New York Office 1".

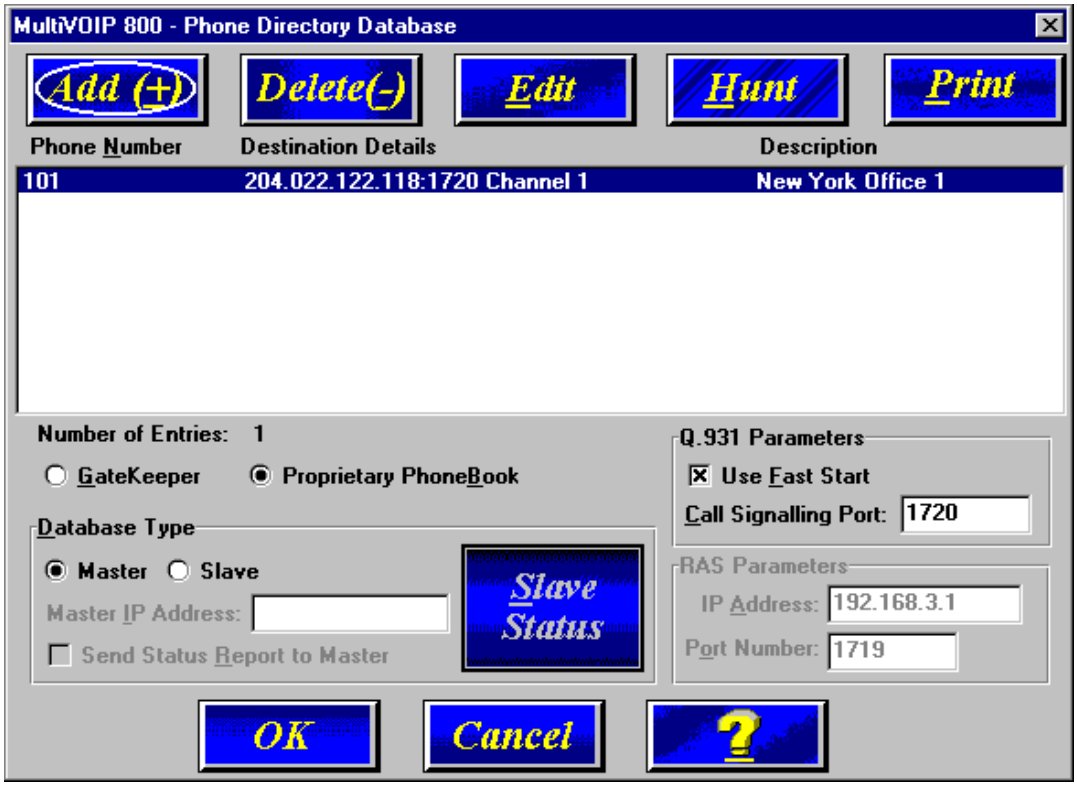
4. The **Station Identification** group includes a **Hunt Group** list. This list enables you to indicate which Hunt Group you want the phone number to use. Or, you can select NO HUNT if you do not want this entry to participate in hunting.

   Note: Hunting is a series of telephone lines organized in such a way that if the first line is busy the next line is hunted and so on until a free line is found. For this example, assign the phone entry to HUNT GROUP #1.

   Once you have assigned this entry to a Hunt Group (or NO HUNT), enter the IP Address of the master MultiVOIP 400 in the **IP Address** box. For example, "204.022.122.118".

   Note: The **Port** box becomes active as you begin to enter the IP Address. The default entry (1720) is the H.323 industry standard Port value used to communicate with other H.323 endpoints.

5. Click **OK** to return to the **Phone Directory Database** dialog box. It now includes phone number (101), destination details (204.022.122.118 Channel 1), and description (New York Office 1).



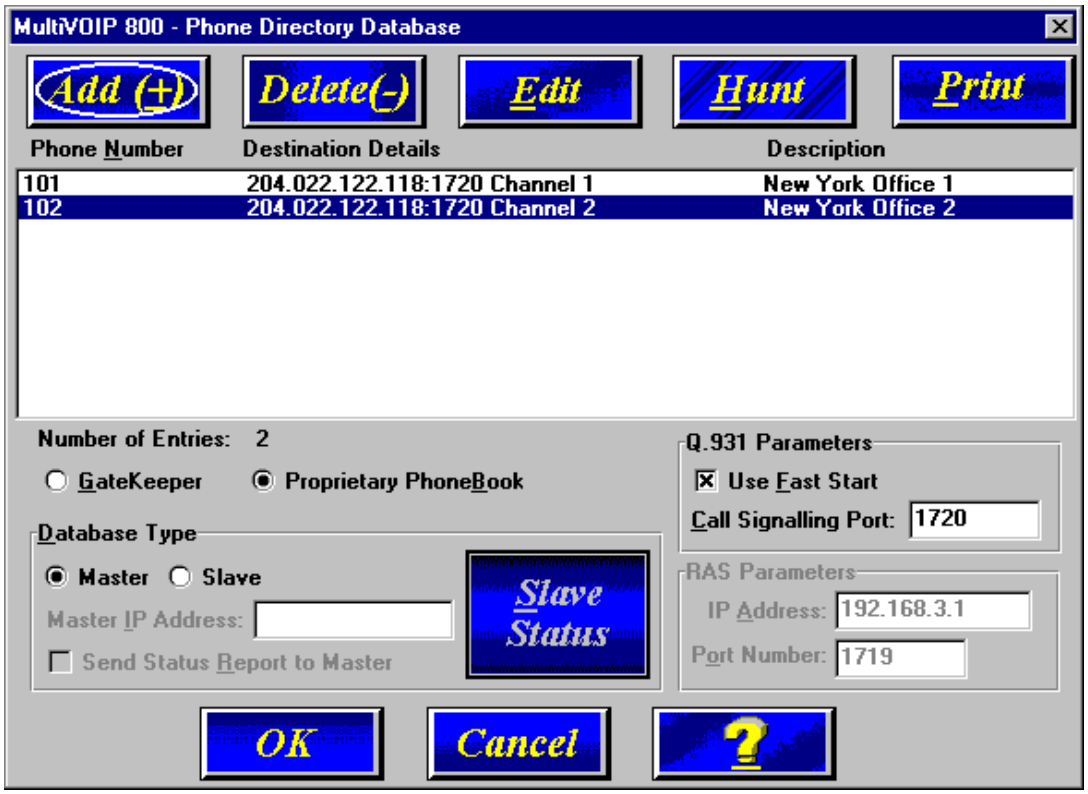
6. To configure Channel 2 on the master MultiVOIP 400, click **Add** and the **Add/Edit Phone Entry** dialog box displays again.

7. Enter the phone number for the MultiVOIP 400 in the **Station Information** group **Phone Number** box. For example, “102”.

8. Enter a description in the **Description** box for the remote MultiVOIP 400 phone number for Channel 2. For example, “New York Office 2”. Then enter the voice channel number in the **Voice Number** box.

9. In the **Station Identification** group, select HUNT GROUP #1 from the **Hunt Group** list. Enter the New York Office 2 IP Address (204.022.122.118). Accept the H.323 industry standard Port value (1720) used to communicate with other H.323 endpoints.

10. Click **OK** and you are returned to the **Phone Directory Database** dialog box which now includes the second number and related information in the **Phone Number** list.



Note: When entering multiple channels, you can also click **Copy From** in the **Add/Edit Phone Entry** dialog box. Enter the phone number you want to copy, click **OK** in the **Copy From** dialog box. Change the information as needed and click **OK** in the **Add/Edit Phone Entry** dialog box.

Both voice channnels belong to Hunt Group #1. If a call from an H.323 endpoint (a MultiVOIP 400 or a stand-alone H.323 endpoint) to Phone Number 101 is unable to be connected, it will automatically connect to the next available phone number in Hunt Group #1, such as Phone Number 102.

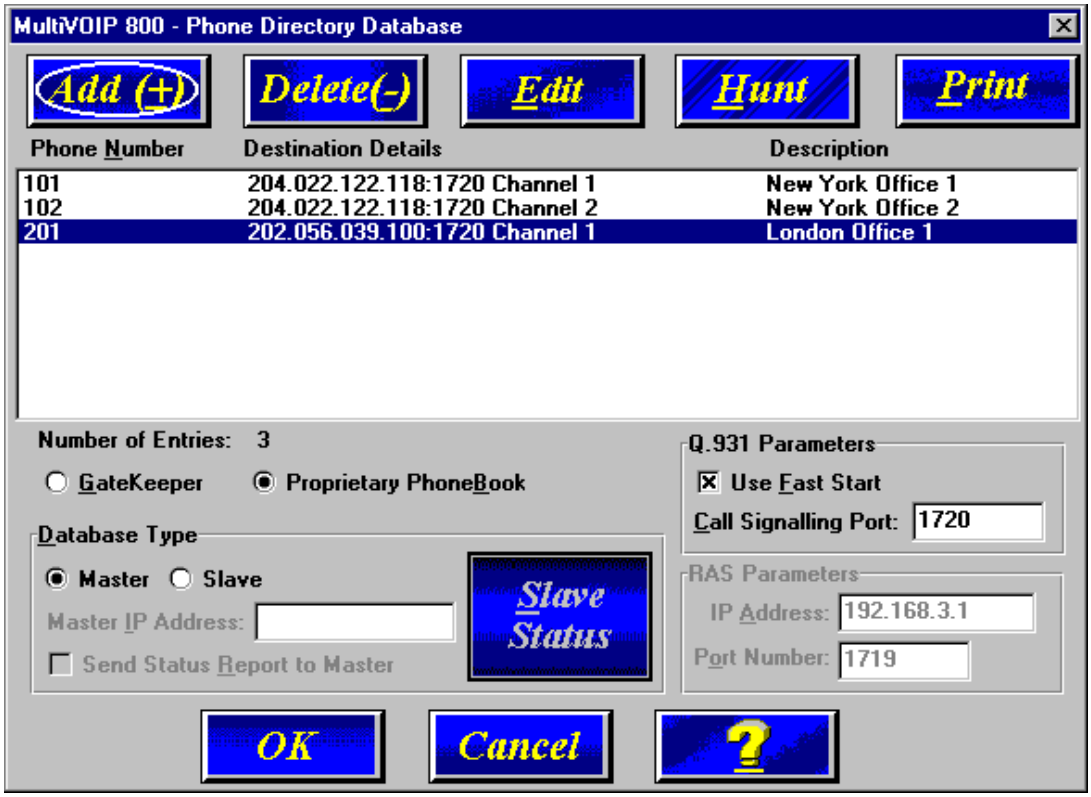
11. Add all other phone numbers (slave units and stand-alone units) to the Phone Directory database. To add Channel 1 of the slave MultiVOIP 400, click **Add**. The **Add/Edit Phone Entry** dialog box displays again.

12. Enter the phone number for the slave MultiVOIP 400 in the **Phone Number** box. For example, “201”.

13. Enter a description in the **Description** box for the remote MultiVOIP 400 phone number for Channel 1. For example, “London Office 1”.

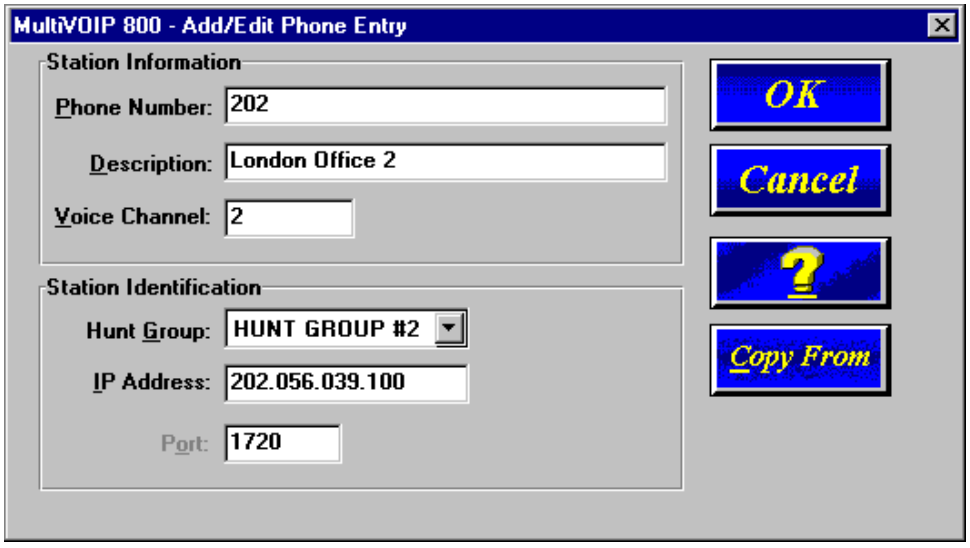
14. In the **Station Identification** group, select **HUNT GROUP #2** from the **Hunt Group** list. Enter London Office 1’s IP Address (202.056.039.100), and accept the H.323 industry standard Port value (1720) used to communicate with other H.323 endpoints.

15. Click **OK** and you are returned to the **Phone Directory Database** dialog box which now includes the remote phone number and related information in the **Phone Number** list.

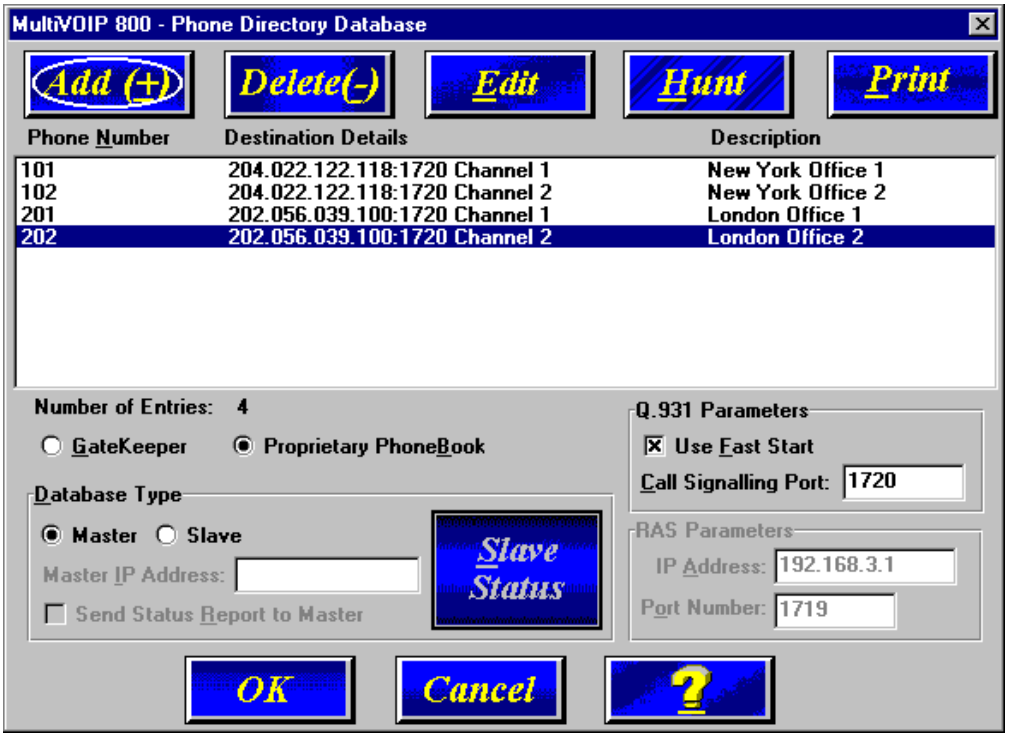


16. To add Channel 2 of the slave MultiVOIP 400, click **Add**. The **Add/Edit Phone Entry** dialog box displays again.
17. Enter the phone number for the remote (slave) MultiVOIP 400 in the **Station Information** group **Phone Number** box. For example, “202”.
18. Enter a description in the **Description** box for the remote MultiVOIP 400 phone number for Channel 2. For example, “London Office 2”.
19. In the **Station Identification** group, select **HUNT GROUP #2** from the **Hunt Group** list, enter the London Office 2’s IP Address (202.056.039.100), and accept the H.323 industry standard Port value (1720) used to communicate with other H.323 endpoints.

Note: Depending on your requirements, you may want calls that cannot make a connection to London Office 1 (Hunt Group #2) to roll over to the New York office instead. In this case, configure that phone entry as a member of Hunt Group #1.



20. Click **OK** and you are returned to the **Phone Directory Database** dialog box which now includes the remote phone number and related information in the **Phone Number** list.



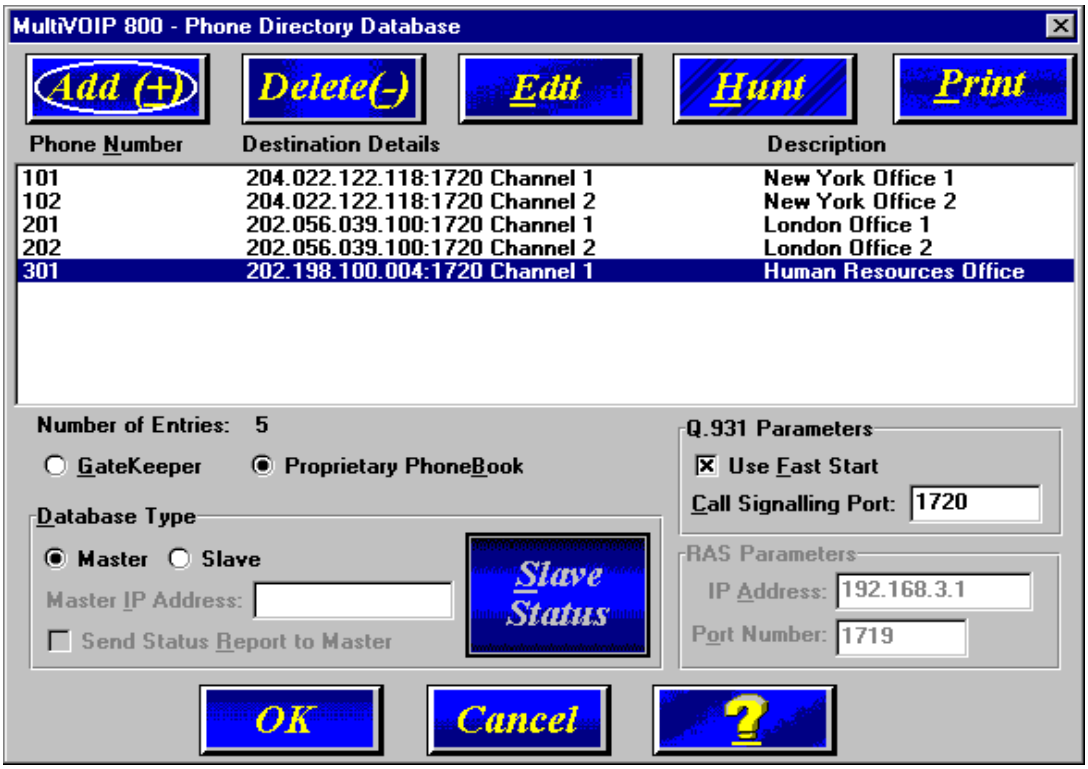
21. To configure a stand-alone endpoint (a PC with Netmeeting software), click **Add**. The **Add/Edit Phone Entry** dialog box displays again.
22. Enter the phone number for the stand-alone endpoint in the **Station Information** group **Phone Number** box. For example, “301”.
23. Enter a description in the **Description** box for the remote MultiVOIP 400 phone number. For example, “Human Resources Desk”.

Note: Because the H.323 endpoint is not a MultiVOIP 400, the Phone Directory database ignores the value in the **Voice Channel** box.

24. In the **Station Identification** group, select **NO HUNT** from the **Hunt Group** list. Enter the Human Resource Desk's IP Address (202.198.100.04) and accept the H.323 industry standard Port value (1720) used to communicate with other H.323 endpoints.

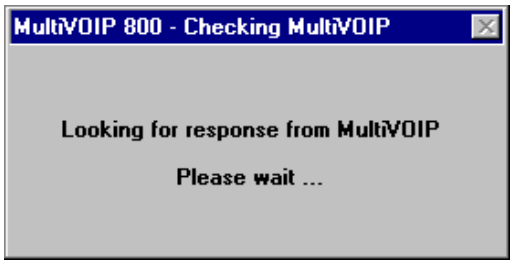
    Note: This stand-alone was not configured as part of a Hunt Group. However, depending on your requirements, you could configure a stand-alone to be part of a Hunt Group.

25. Click **OK** and you are returned to the **Phone Directory Database** dialog box which now includes the stand-alone phone number and related information in the **Phone Number** list.

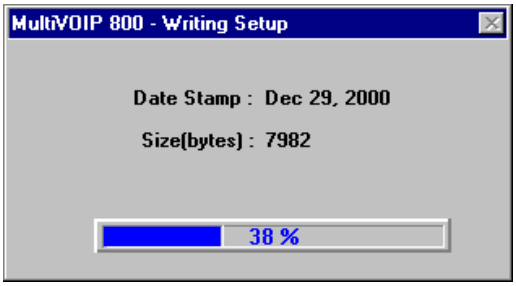


26. When you have finished, click **OK** to download the setup configuration to the MultiVOIP 400.

27. The **Checking MultiVOIP** dialog box displays.



28. Once the setup program receives a response from the MultiVOIP 400, the **Writing Setup** dialog box displays, indicating that the setup configuration is being written to the MultiVOIP 400.



29. After the setup has been written to the MultiVOIP 400, the unit is rebooted.

30. Check to ensure that the **BTG** LED on the MultiVOIP 400 is off after the download is complete. This may take several minutes while the MultiVOIP 400 reboots.

31. You are returned to the MultiVOIP MVP400 screen frtom which you can load Acrobat Reader to your PC. This allows you to view and/or print the User Guide by clicking the **Install Manuals** or **View Manuals** icon.

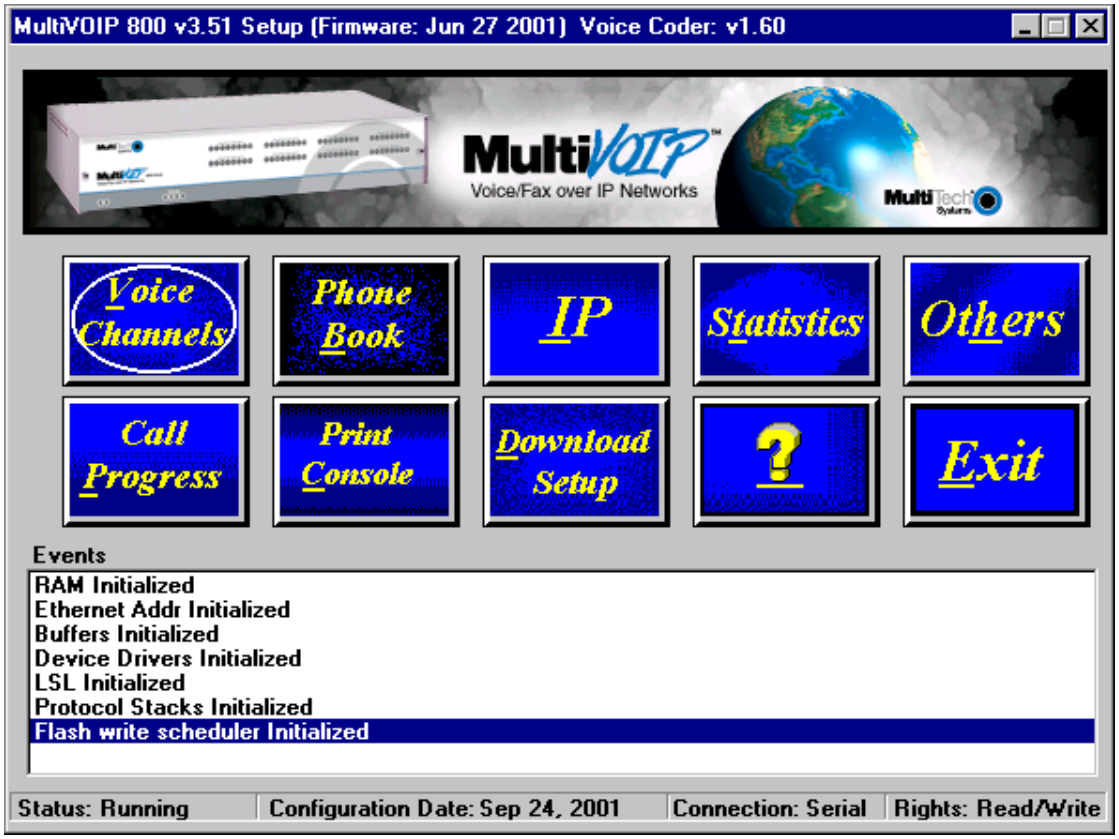
At this time your master MultiVOIP 400 is configured. Proceed to the “Configuring Your Slave MultiVOIP 400s” section.

Configuring Your Slave MultiVOIP 400s

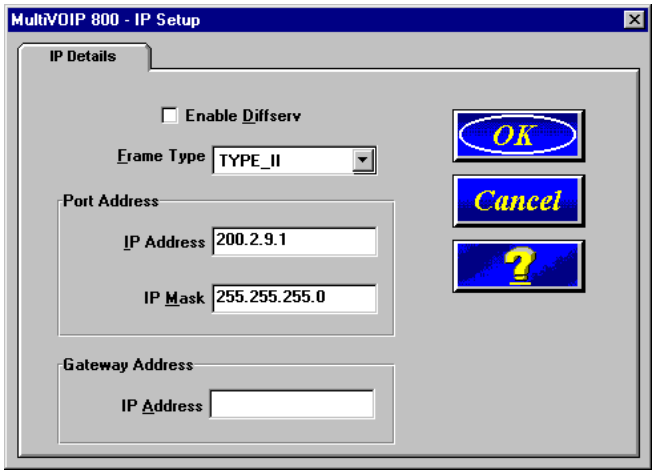
If the **Proprietary Phonebook** option is selected in the **Phone Directory Database** dialog box, then you will need to configure all remote H.323 endpoints as slave units. For example, the MultiVOIP 400 at the company's subsidiary office in London would need to be configured as a slave.

CAUTION: If you are installing a MultiVOIP 400 behind a Firewall, the Firewall must support H.323. Refer to your Firewall user documentation to enable H.323 support.

1. Disconnect the PC from the Command port of the master MultiVOIP 400 and connect it to the Command port on the slave MultiVOIP 400.
2. From your desktop, click **Programs I MultiVOIP 800 I MultiVOIP 800 Configuration**. The Main menu displays.



3. Click **IP** to display the **IP Setup** dialog box.



Select the **Enable Diffserv** check box to enable Differentiated Services on routers that support this service. This feature provides priority to voice packets so that they are not delayed because of large data files being downloaded.

The default **Frame Type** is **TYPE_II**. If this does not match your IP network, change the Frame Type by selecting it from the **Frame Type** list. The Frame Type choices are **TYPE_II** and **SNAP**.

4. In the **Port Address** group, enter the **IP Address** and **IP Mask**. In the **Gateway Address** group, enter the gateway **IP Address**.

   The IP Address is the unique IP address that you assign to the MultiVOIP 400, and the Gateway Address is the IP address of the device (such as a network router) connected to the Internet/ Intranet.

   Click **OK** when you are finished. The Main menu displays.

5. From the Main menu, click **Voice Channels** to display the **Channel Setup** dialog box. The **Channel Setup** dialog box displays. The four tabs in this dialog box define the channel interface, voice/fax parameters, Billing/Misc parameters, and regional telephone parameters for each channel.

Configure each channel for the type of interface you are using. **Channel 1** displays by default on the **Interface** tab. To change the channel number, select the channel you want to configure from the **Select Channel** list.

Feature options are enabled or unavailable according to the selected interface type. The **Inter Digit Time** check box is available for all interface types. This option defines the maximum amount of time that the unit will wait before mapping the dialed digits to an entry in the Phone Directory Database. If too much time elapses between digits, and the wrong numbers are mapped, you will hear a rapid busy signal. If this happens, hang up and dial again. The default is **2** seconds.

6. The **Interface** group defaults to **FXS (Loop Start)**. Select the interface option that corresponds to the interface type being connected to the Voice/Fax Channel 1 jack on the back panel of the MultiVOIP 400.

FXS (Loop Start): If a station device such as an analog telephone, fax machine, or KTS (Key Telephone System) is connected to the Voice/Fax connector on the back of the unit, **FXS (Loop Start)** will likely be the correct Interface option.

FXS (Ground Start): If the station device uses ground start, then choose the **FXS (Ground Start)** option. Refer to the device's user documentation.

For both FXS Loop Start and FXS Ground Start , the **Ring Count FXS** window allows you to set the maximum number of rings output on the FXS interface before hanging up and releasing the line to another call. The default setting is **8** counts.

Note: Zero (0) means no rings. The caller hears a busy tone.

FXO: If you are using an analog extension from your PBX, then select the **FXO** option. Check with your in-house phone personnel to verify the connection type.

If FXO is selected, the **Dialing Options Regeneration, Flash Hook Timer,** and **Ring Count** groups are enabled. Check with your in-house phone personnel to verify whether your local PBX dial signaling is **Pulse** or **DTMF** (tone). Then, set the Regeneration option accordingly. The **Flash Hook Timer** allows you to enter the time, in milliseconds, for the duration of the flash hook signal output on the FXO interface. The default setting is **600** milliseconds. The **Ring Count FXO** window allows you to set the number of rings received on the FXO interface before the MultiVOIP 400 answers the incoming call. The default setting is **2** counts.

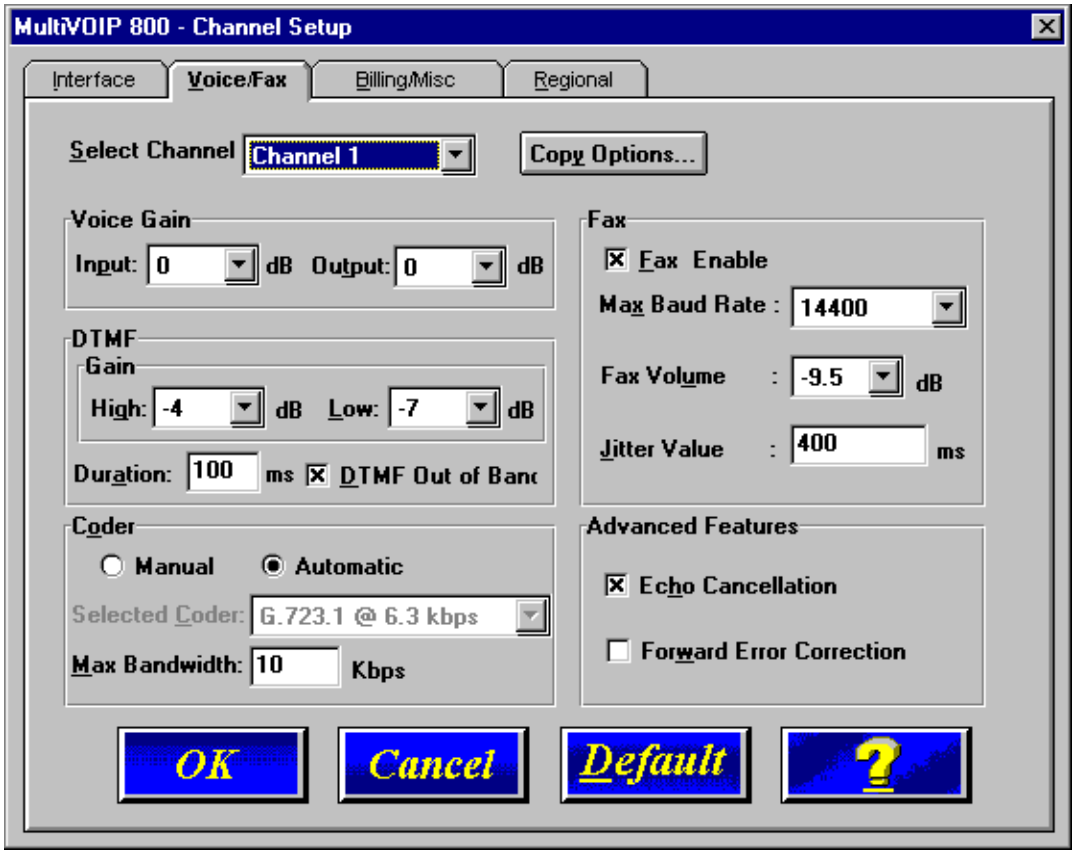
Note: Zero (0) means that the MultiVOIP 400 never answers.

For FXO-to-FXO communications, you can enable a specific type of FXO Disconnect: **Current Loss**, **Tone Detection**, or **Silence Detection**. Check with your in-house phone personnel to verify the preferred type of disconnect. Enabling **Tone Detection** activates the Disconnect Tone Sequence options. For **Disconnect Tone Sequence**, you can select either one or two tones that will disconnect the line. The person hanging up a call must then hit the key or keys that will produce those tones. For **Silence Detection**, select **One Way** or **Two Way**, then set the timer for the number of seconds of silence before disconnect. The default value of **15** seconds may be shorter than desired for your application.

E&M: If you are connecting to an analog E&M trunk on your PBX, then choose the E&M interface option to enable the E&M Options group. Check with your in-house phone personnel to determine if the signaling is **Dial Tone** or **Wink** and if the connection is 2-wire or 4-wire. If Wink signaling is used, then the **Wink Timer** is enabled with a default of **250** milliseconds. The range of the Wink Timer is from 100 to 350 milliseconds. Check with your in-house phone personnel for this timer setting.

Note: After configuring a given channel, you can copy that channel configuration by clicking **Copy**. Everything on the **Interface** tab will be copied to the other channel.

7. Repeat the above step to configure the interface type for voice/fax channel 2.
8. The **Voice/Fax** tab displays the parameters for the voice gain, DTMF (Dual Tone Multi-Frequency) gain, voice coder, faxing, and advanced features such as Silence Compression, Echo Cancellation, and Forward Error Correction.



9. You can set up the input and output voice gain so that the volume can be increased or decreased. Input gain modifies the level of the audio coming in to the voice channel before it is sent over the Internet to the remote MultiVOIP 400. Output gain modifies the level of the audio being output to the device attached to the voice channel. Make your selections from the **Input** and **Output** lists in the **Voice Gain** group. The valid range is +31dB to –31dB with a recommended default value of **0**.

   The **DTMF Gain** group controls the volume level of the digital tones sent out for touchtone dialing. The selections from the **Gain High** and **Gain Low** lists control the gain in decibels of the high and low tones in the tone pairs. The default gain values are -4 dB and -7 dB respectively. The DTMF gain should be changed only with the assistance of Multi-Tech Technical Support.

   When the **DTMF Out of Band** check box is selected, the MultiVOIP reproduces DTMF tones rather than passing them from input to output.

10. To change the voice coder, select the channel from the **Select Channel** list, then click **Manual** in the **Coder** group. To select the appropriate coder, select your new voice coder entry from the **Selected Coder** list.

    If you changed the voice coder, ensure that the same voice coder is used on the voice/fax channel you are calling. Otherwise, you will always get a busy signal.

    Note: If you allow the coder to be selected automatically, enter the **Max Bandwidth**. Check with your VOIP administrator to determine how much bandwidth is available.

11. The **Fax** group enables you to send/receive faxes on the selected voice/fax channel. You can select the maximum baud rate for faxes, select the fax volume, and enter the jitter value in milliseconds.

    When receiving fax packets from a remote MultiVOIP 400, it is possible for individual packets to be delayed or received out of order due to traffic conditions on the network. To compensate for this effect, the MultiVOIP 400 uses a Jitter Buffer. The **Jitter Value** allows the MultiVOIP 400 to

wait a user-definable period of time, in milliseconds, for delayed or out-of-order fax packets. The range of allowable Jitter Values is 0 to 400 milliseconds with a default of **400** milliseconds.

If you do not plan to send or receive faxes on a given voice/fax channel, you can disable faxes in the **Fax** group.

12. You can enable the voice/fax advanced features by selecting the **Silence Compression**, **Echo Cancellation**, or **Forward Error Correction** check boxes.

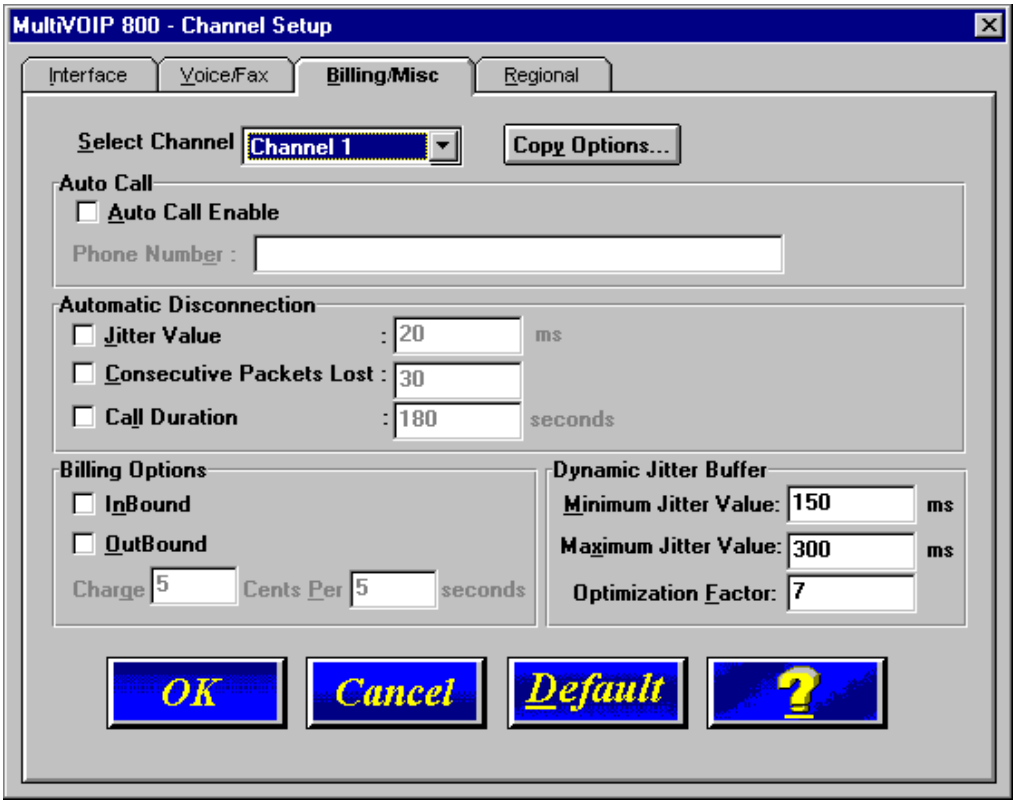
    The **Silence Compression** check box defines whether silence compression is enabled for this voice channel. If silence compression is enabled, the MultiVOIP 400 will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel.

    The **Echo Cancellation** check box defines whether echo cancellation is enabled for this voice channel. If echo cancellation is enabled, the MultiVOIP 400 will remove echo delay, which improves the quality of sound.

    The **Forward Error Correction (FEC)** check box defines whether forward error correction is enabled for this voice channel. The FEC feature allows some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel.

    Note: After configuring a given channel, you can copy that channel configuration by clicking **Copy**. Everything on the **Voice/Fax** tab will be copied to the other channel.

    The **Billing/Misc** tab displays the parameters for auto call, automatic disconnection, billing options, and dynamic jitter buffer.



13. If you want to dedicate a local voice/fax channel to a remote voice/fax channel (so you will not have to dial the remote channel), click the **Auto Call Enable** check box in the **Auto Call** group. Then enter the phone number of the remote MultiVOIP 400 in the **Phone Number** box.

14. The **Automatic Disconnection** group provides three options to be used singly or in combination.

The **Jitter Value** defines the average inter-arrival packet deviation (in milliseconds) before the call is automatically disconnected. Jitter is the inter-arrival packet deviation (phase shift of digital pulses) over the transmission medium that causes voice breakup which can be particularly disruptive to voice communications. The default setting is 20 milliseconds. A higher value means that the voice transmission will be more accepting of jitter. A lower value will be less tolerant of jitter.

Consecutive Packets Lost defines the number of consecutive packets that are lost after which the call is automatically disconnected. The default setting is **30** packets.

Call Duration defines the maximum length of time (in seconds) that a call remains connected before the call is automatically disconnected. The default setting is **180** seconds. A call limit of three minutes may be too short for most configurations. Therefore, you may want to increase this default value.

15. You can set billing options for inbound and/or outbound calls by checking them in the **Billing Options** group and then entering the charge in cents per number of seconds.

16. A minimum and maximum set of values can be set for **Dynamic Jitter Buffer**. When receiving voice packets from a remote MultiVOIP 400, it is possible to experience varying delays between packets due to traffic conditions on the network. This is called Jitter. To compensate for this effect, the MultiVOIP 400 uses a Dynamic Jitter Buffer. The Jitter Buffer allows the MultiVOIP 400 to wait for delayed voice packets by automatically adjusting the length of the Jitter Buffer between configurable minimum and maximum values. An **Optimization Factor** adjustment controls how quickly the length of the Jitter Buffer is increased when jitter increases on the network. The length of the jitter buffer directly effects the voice delay between MultiVOIP 400 gateways.

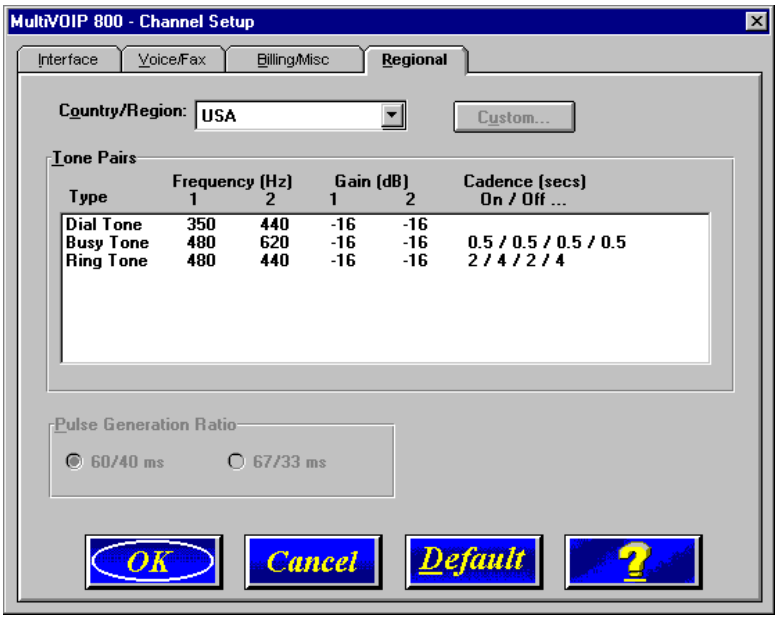
    The Minimum Jitter Value default setting is **150** milliseconds, the Maximum Jitter Value default setting is **300** milliseconds, and the Optimization Factor default setting is **7**.

    Note: After configuring a given channel, you can copy that channel configuration by clicking **Copy**. Everything on the **Billing/Misc** tab will be copied to the other channel.

    If your country/region is not USA, the default , click the **Regional** tab and proceed to step 17. Otherwise, proceed to step 18 to begin building your phone directory database.

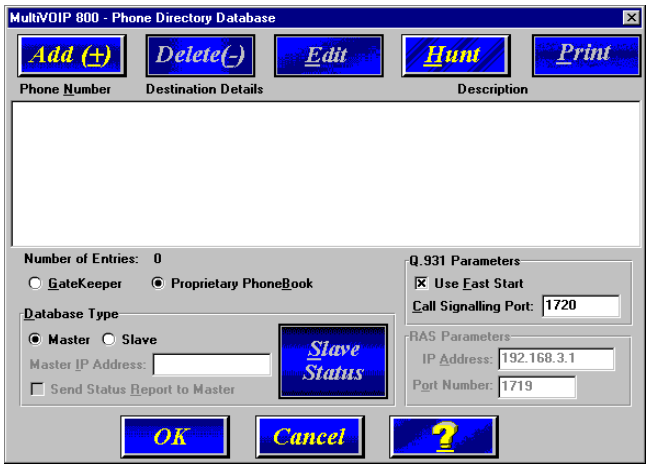
17. To change the Tone Pairs on the **Regional** tab, select your specific country or region from the **Country/Region** list.

    Note: If your country or region is not listed, click **Custom** to define it.



The **Tone Pairs** group enables you to select and modify the parameters. Click **OK** when finished and proceed to step 6 to begin building your phone directory database.

18. From the Main menu, click **Phone Book** to display the **Phone Directory Database** dialog box. Verify that the **Proprietary Phonebook** option is selected and in the **Database Type** group, select the **Slave** option. The **Master IP Address** box becomes active.



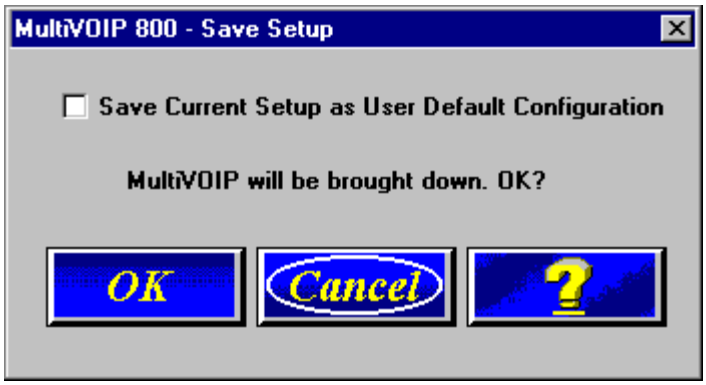
Note: After you have selected the **Slave** option, the **Slave Status** button is replaced by the **Update** button. Once your Phone Directory database has been established, you can click this button to refresh the entries in the Phone Directory Database window.

19. Enter the IP address (204.022.122.118) of the New York Office MultiVOIP 400 in the **Master IP Address** box and select the **Send Status Report to Master** check box. This establishes the New York office MultiVOIP 400 as the master.

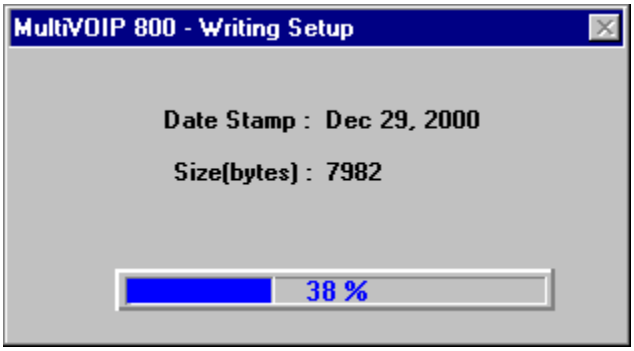
    Note: In a Dial-On-Demand (DOD) network, you should leave the **Send Status Report to Master** check box cleared. This enables the router to disconnect whenever there is no voice activity. Note that slaves with the **Send Status Report to Master** check box cleared will show up as "Unknown" when viewing slave status on the master.

20. Click **OK** to return to the Main menu.

21. Click **Download Setup** to write the new configuration to the slave unit. The **Save Setup** dialog box displays.



22. Select the **Save Current Setup as User Default Configuration** check box and click **OK**. The **Writing Setup** dialog box displays as the setup configuration is written to the MultiVOIP 400.



After the setup is written to the MultiVOIP 400, it reboots.

23. Verify that the **BTG** LED on the MultiVOIP 400 is off after the download is complete. This may take several minutes as the MultiVOIP 400 reboots.
24. You are returned to the Main menu.

    Your MultiVOIP 400 is operational at this time.

Repeat the process for each of the slave units. When all slaves have been configured, go to the “Deploying the Network” section.

Deploying the VOIP Network

With the **Proprietary Protocol** option enabled on the **Phone Directory Database** dialog box, the VOIP Administrator must develop the VOIP Dialing Directory and deploy the pre-configured slave MultiVOIP 400s to their remote sites. The remote site administrators need only connect power to the pre-configured MultiVOIP 400, connect the MultiVOIP 400 to their Ethernet LAN and predefined telephone equipment, and then wait for the phone directory database to be downloaded.

With the **Gatekeeper** option selected on the **Phone Directory Database** dialog box, all MultiVOIP 400s are configured as master and cannot be downloaded. In this case, each MultiVOIP 400 Phone Book will be programmed with phone numbers for its own channels. These phone numbers are remotely registered with the H.323 Gatekeeper. See the “Registering with a Gatekeeper Phone Directory” section for more information.

Remote Site Administrator

The following steps are for MultiVOIP 400 H.323 endpoints. For non-MultiVOIP 400 H.323 endpoints, refer to the appropriate installation documentation.

1. Unpack your MultiVOIP 400.
2. Connect one end of the power supply to a live AC outlet and connect the other end to the **Power** connection on your MultiVOIP 400 (See Figure 3-1).

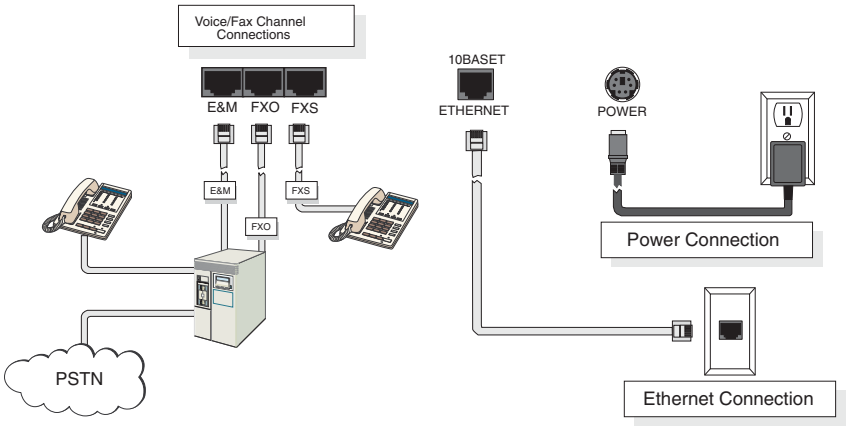


Figure 3-1. Remote Site Cable Connection

3. Connect a network cable to the **ETHERNET 10BASET** (RJ-45) connector on the back of your MultiVOIP 400.
4. If you are connecting a station device such as an analog telephone, a fax machine, or a Key Telephone System (KTS) to your MultiVOIP 400, connect the smaller end of a special adapter cable (supplied) to the **Voice/Fax Channel 1 FXS** connector on the back of the MultiVOIP 400 and the other end to the station device.

   If you are connecting a PBX extension to your MultiVOIP 400, connect the smaller end of a special adapter cable (supplied) to the **Voice/Fax Channel 1 FXO** connector on the back of the MultiVOIP 400 and the other end to the PBX extension.

   If you are connecting an E&M trunk from a telephone switch to your MultiVOIP 400, connect one end of an RJ-45 phone cord to the **Voice/Fax Channel 1 E&M** connector on the back of the unit and the other end to the trunk phone jack.

   If you are connecting to an E&M trunk, ensure that the E&M trunk jumper is in the correct position for the E&M type trunk. The default E&M jumper position is E&M type 2. To change the E&M jumper position, perform the E&M jumper block positioning procedure in Chapter 2.

5. Repeat the above step to connect the remaining telephone equipment to each Voice/Fax Channel on your MultiVOIP.
6. Turn on power to the MultiVOIP 400 by placing the ON/OFF switch on the back panel to the ON position. Wait for the BTG LED on the MultiVOIP 400 to go off before proceeding. This may take a couple of minutes.
7. At this time your VOIP network should be fully operational. Dial one of the sites in your network using the dialing directory supplied by your network administrator.

MultiVOIP™

Voice/Fax over IP Networks

Chapter 4 - MultiVOIP 400 Software

MultiTech® Systems

Introduction

This chapter describes various features of the MultiVOIP 400 software that enable you to change (update) the configuration of your MultiVOIP 400. The basic configuration parameters were established during the loading of the software (Chapter 3). The MultiVOIP 400 software and configuration utilities described in this chapter enable you to change that initial configuration as necessary.

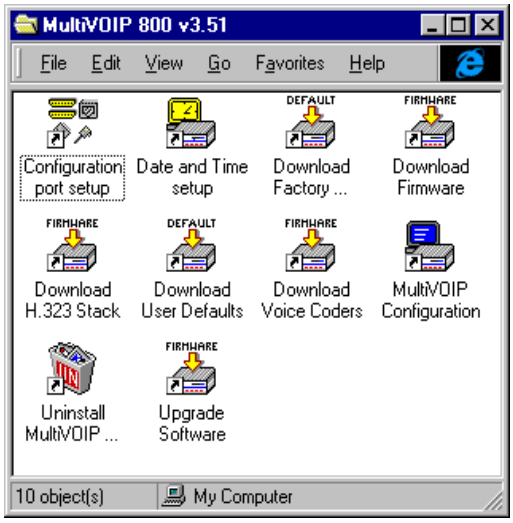
The primary interface to the MultiVOIP 400 software is the Main menu (**MultiVOIP 800 Setup** is on the title bar) which includes individual buttons that enable you to quickly and easily select a desired function. These features are discussed in detail in the MultiVOIP 400 Configuration section later in this chapter.

The MultiVOIP 800 Configuration (Main menu) utility along with eight other configuration utilities provide full software functionality for your MultiVOIP 400. The **Configuration Port Setup** utility enables you to change the method by which you access the MultiVOIP 400, whether through a direct connection of a PC to the Command Port on the MultiVOIP 400, or via your Internet or LAN connection to the LAN port on the MultiVOIP 400. The **Date and Time Setup** utility enables you to easily set the date and time used for data logging in the MultiVOIP 400. The **Download Factory Defaults** utility enables you to return the configuration to the original factory settings. The **Download Firmware** utility enables you to download new versions of firmware as enhancements become available. The **Download User Defaults** utility enables you to repeat the download user defaults process (part of software installation) and update the MultiVOIP 400 configuration with any necessary changes. The **Download Voice Coders** utility enables you to download voice coders to the MultiVOIP 400 after repair or upgrade. The **Download H.323 Stack** utility enables you to download the H.323 protocols to the MultiVOIP 400 after repair or upgrade. The **Uninstall MultiVOIP 800 Configuration** utility removes most of the MultiVOIP 400 software from your PC. The **Upgrade Software** utility enables you to install updated versions of the MultiVOIP 400 software.

The MultiVOIP 400 software includes context-sensitive Help. The Help contains definitions and recommended values for each dialog box or menu.

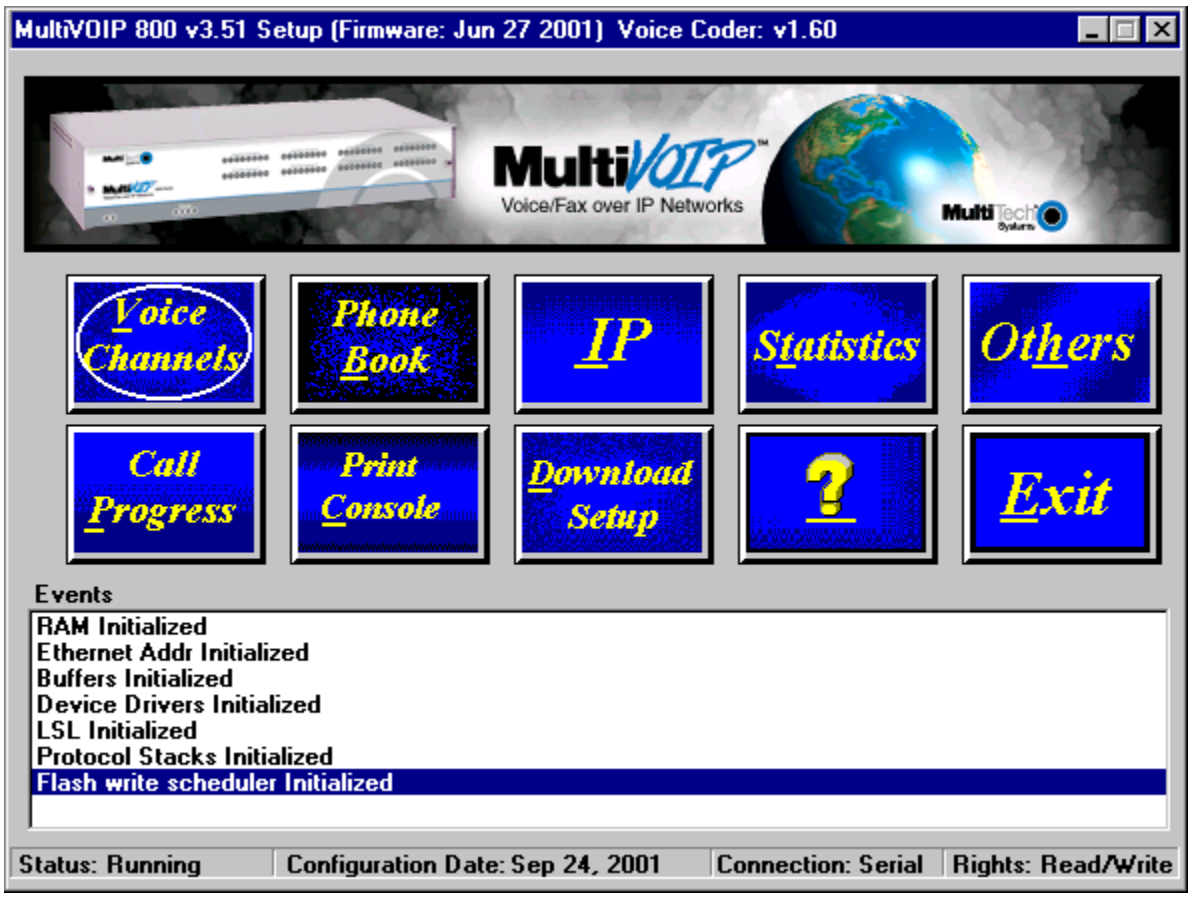
Before You Begin

The MultiVOIP 400 software operates in a Microsoft Windows environment. The MultiVOIP 400 program group contains icons for all the utilities described above. In Windows 2000/NT/98/95, you can access the individual utility programs either by clicking **Start | Programs | MultiVOIP 800 v3.51| *(utility)***, or by double-clicking the utility icon in the MultiVOIP 400 program group.



MultiVOIP 400 Configuration

To configure your MultiVOIP, double-click the **MultiVOIP Configuration** icon in the **MultiVOIP 400** program group or in Windows 2000/NT/98/95, click **Start** | **Programs** | **MultiVOIP 800 v3.51** | **MultiVOIP Configuration**. The **MultiVOIP 800 Setup** menu consists of 10 buttons, an Events window in the middle of the menu, and a status bar at the bottom of the menu. The 10 buttons enable you to display and change the voice channels and IP protocol parameters, display and manage the Phone Book listing, view statistics and call progress, and change features such as SNMP Agent, Telnet Server, WEB Server, and assign a MultiVOIP 400 password.

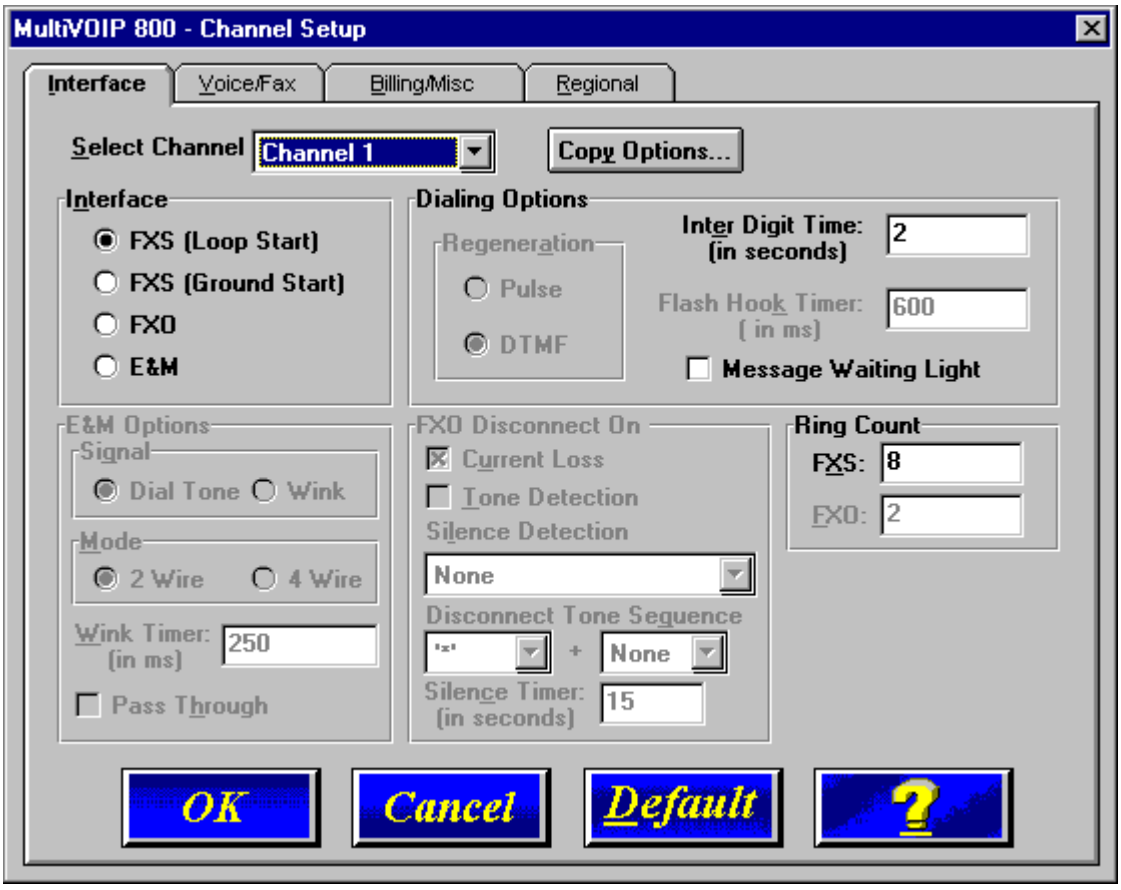


The Events window in the lower third of the Setup menu provides information about the boot process.

The status bar at the bottom of the Setup menu displays the current status of the unit and shows, for example, if it is running, the most recent date the unit was configured, the type of connection you have to the unit, and your rights. It shows if your PC is connected directly to the command port of the MultiVOIP 400 or is communicating with the Ethernet port. The last box on the status bar is the Rights box which displays either Read/Write or Read Only rights. The first user to communicate with the MultiVOIP 400 has Read/Write rights that enable the user to view and/or change the configuration of the MultiVOIP 400. Additional users have Read Only rights and can only display the configuration of the MultiVOIP 400, but are prohibited from changing the configuration.

Changing Channel Parameters

The channel parameters include the interface type and its options, voice and fax settings, billing and security, and voice communications for the country and region in which the MultiVOIP 400 is operating. The **Channel Setup** dialog box, accessed by clicking **Voice Channels** on the Setup menu, has four tabs that display the following categories of channel information -- Interface, Voice/Fax, Billing/Misc, and Regional.



Interface

The **Interface** tab defines the parameters related to the physical interface of the voice/fax channel. Depending on the interface type selected (FXS, FXO, or E&M), other options on the **Interface** tab will be grayed out (become inactive) indicating that they do not apply to the selected interface. The Inter Digit Time applies to all interface types.

The **Inter Digit Time** box in the **Dialing Options** group defines the amount of time in seconds that the MultiVOIP 400 waits between digits as they are entered by the user. If this timer expires, the MultiVOIP 400 will immediately attempt to match the digits entered to an entry in the Phone Directory Database. The range for this option is 2 to 100 with a default of 2.

If the interface type is FXO, the **Regeneration** group in the **Dialing Options** group defines how the MultiVOIP 400 recreates telephone numbers that were detected at the remote end. You can select Pulse (for rotary dial telephones) or DTMF Tone dialing (touch-tone), depending upon the dialing type that is supported by the PBX or exchange. When FXO is the Interface, the **Flash Hook Timer** box is enabled (activated) Enter the time, in milliseconds, for the duration of the flash hook signals output on the FXO interface. The default is **600** milliseconds**.**

FXS Interface

The FXS Interface is used to connect telephones, fax machines, key telephone systems, and so on to the MultiVOIP 400. In addition, you need to select either Loop Start or Ground Start. Most of the equipment mentioned will use Loop Start, which is the default.

FXO Interface

The FXO Interface is used to connect PBX extensions or central office telephone lines. You also, need to select DTMF or Pulse dialing in the **Regeneration** box in the **Dialing Options** group. If you are unsure of the correct selection, contact the personnel in charge of your PBX or your local telephone company to determine whether pulse or DTMF should be used.

E&M Interface

The E&M Interface is used to connect PBX E&M trunks. You will need to select between Dial Tone or Wink signaling and also between 2-wire and 4-wire mode. If wink signaling is selected, the wink timer box becomes active with a range from 100 to 350 milliseconds. Contact the personnel in charge of your PBX to determine the proper configuration of these settings.

FXO Disconnect On

The **FXO Disconnect On** option applies when two MultiVOIP 400s are used in an FXO-to-FXO configuration. When you have an FXO-to-FXO configuration, you need to determine the method of terminating the call. Three methods of terminating the call are provided: **Current Loss**, **Tone Detection**, or **Silence Detection**. **Current Loss** is the preferred method.

Current Loss has to be supported by your PBX or local telephone company. Current Loss terminates the call when the PBX or local telephone company switch detects a person hanging up the phone and opens the local circuit for a minimum of 600 milliseconds.

Tone Detection terminates the call when the party who wishes to disconnect enters a one or two digit sequence on the telephone keypad. Valid digits are zero to nine, *, #, and A thru D.

Note: A through D are extended DTMF tones supported by some PBX or central office equipment and are not the same as letters a-d on the standard telephone key pad.

Silence Detection can be silence in one direction or silence in both directions for a specified amount of time. The amount of time is defined by the entry in the Silence Timer. The range of the Timer is from one to 65535 seconds (roughly 18 hours). The default is 15 seconds.

Ring Count

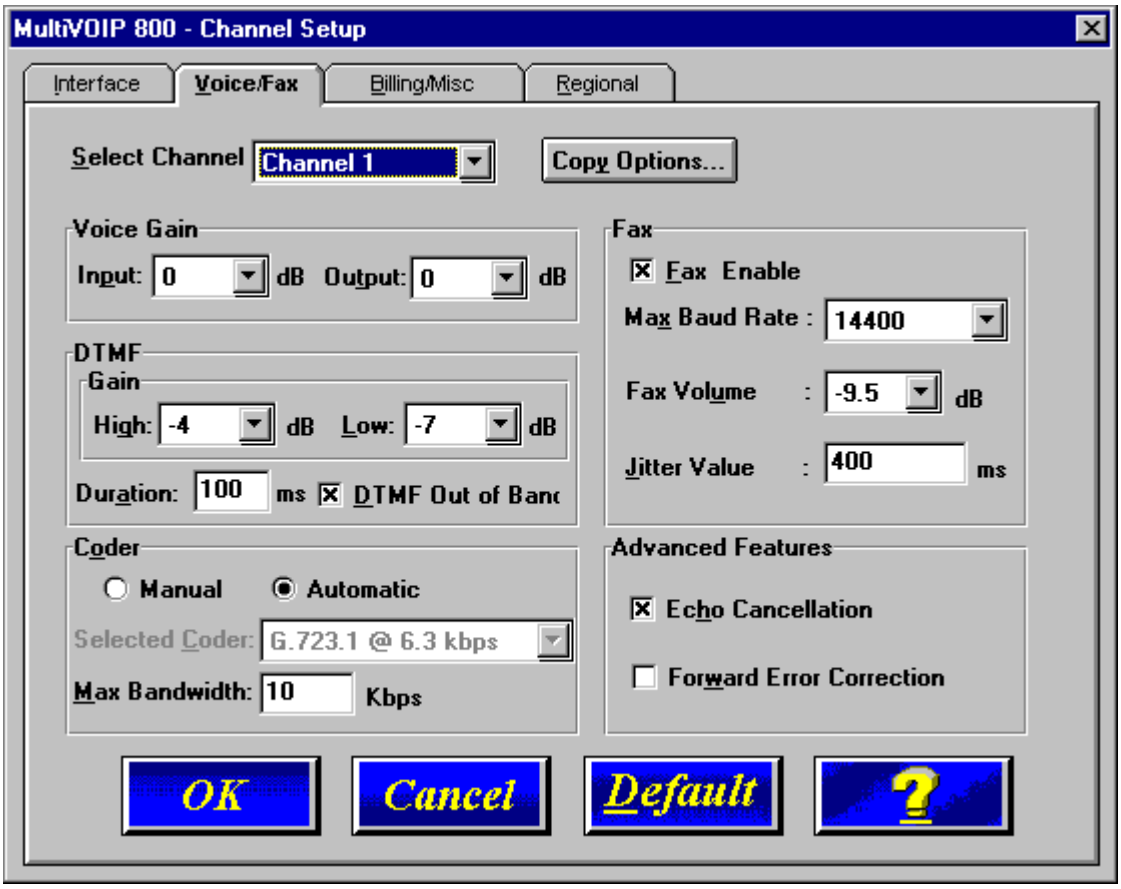
This box enables you to enter the maximum number of rings output on the FXS interface (default is 8) before hanging up and releasing the line to another call or the number of rings (default is 2) that must be received before the FXO port answers an incoming call.

A setting of 0 (zero) on the FXS interface disables the generation of rings. The caller will receive a Busy tone.

A setting of 0 (zero) on the FXO interface causes the FXO port to ignore rings from the attached PBX or exchange, disabling access to the MultiVOIP 400.

Voice/Fax

The Voice/Fax tab controls voice and DTMF gain, voice coder, fax settings, and advanced options.



The **Voice Gain** group enables you to select the Input and Output voice gain. Gain is the increased signaling power that occurs as the signal is boosted by the MultiVOIP 400. The **Input Gain** drop-down list defines the input gain for this voice channel. Before your MultiVOIP 400 digitizes voice, the volume can be increased or decreased. Input gain modifies the level of the audio coming in to the voice channel before it is sent over the Internet to the remote MultiVOIP 400. The valid range for this option is +31dB to –31dB. The recommended and default value is **0**. The **Output Gain** list defines the voice output gain for this voice channel. Before your MultiVOIP 400 converts digital voice back to analog, the volume can be increased or decreased. The output gain modifies the level of the audio being output to the device attached to the voice channel. The valid range for this option is +31dB to –31dB. The recommended and default value is **0**.

The **DTMF Gain** (Dual Tone Multi-Frequency) group controls the volume level of the digital tones sent out for Touch-Tone dialing. The Gain High and Gain Low boxes control the gain in dB (decibels) of the High and Low tones in the tone pairs. The default gain values are **-4** dB and **-7** dB, respectively. DTMF Gain should not be changed except under supervision of MultiTech's Technical Support.

When the **DTMF Out of Band** check box is selected, the MultiVOIP reproduces DTMF tones rather than passing them from input to output.

The MultiVOIP 400 supports many state-of-the art ITU (International Telecommunications Union) voice coders. The **Voice Coder** list enables you to select from a range of coders with specific bandwidths. The higher the bps rate, the more bandwidth is used. The channel that you are calling has to have the same voice coder selected. Otherwise, you will always get a Busy signal.

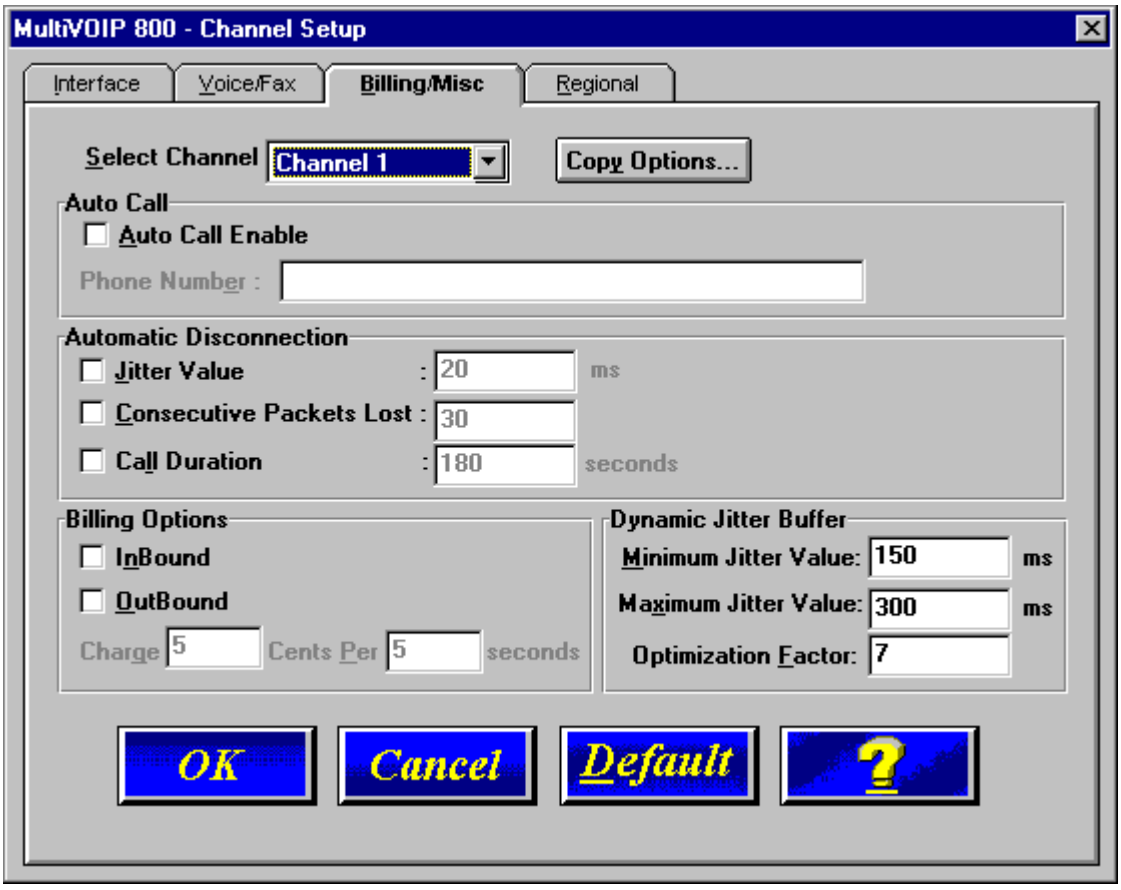
The **Fax** group enables a fax machine to transmit and receive faxes through the MultiVOIP 400. If a

fax machine is connected to one of the voice/fax channels, the **Max Baud Rate** should be set to match the baud rate of the fax machine (refer to user documentation). The **Fax Volume** setting controls the output level of the fax tones. This setting should be changed only under the direction of Multi-Tech's Technical Support personnel (see Chapter 6 - Warranty, Service and Tech Support). The **Jitter Value** setting defines the inter-arrival packet deviation (in milliseconds) for the fax transmission. A higher value will increase the amount of delay, enabling a higher percentage of packets to be reassembled. A lower value would decrease the amount of delay enabing a lower percentage of packets to be reassembled.

The **Advanced Features** group allows you to enable **Silence Compression** so that a MultiVOIP 400 will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel. **Echo Cancellation** for a particular voice channel will remove echo and improve the quality of sound. **Forward Error Correction** enables some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel.

Billing/Misc

This tab controls the parameters for auto call, automatic disconnection, billing options, and dynamic jitter buffer.



The **Auto Call** option enables the local MultiVOIP 400 to call a remote MultiVOIP 400 without the user having to dial a Phone Directory Database number. As soon as you access the local MultiVOIP 400 voice/fax channel, the MultiVOIP 400 immediately connects to the remote MultiVOIP 400 that you identified in the Remote MultiVOIP 400 Phone Number box of this option.

The **Automatic Disconnection** group provides three options which can be used singly or in any combination. The **Jitter Value** defines the average inter-arrival packet deviation (in milliseconds) before the call is automatically disconnected. The default is **20** milliseconds. A higher value means

voice transmission will be more accepting of jitter. A lower value is less tolerant of jitter.

Consecutive Packets Lost defines the number of consecutive packets that are lost after which the call is automatically disconnected. The default is **30** packets.

Call Duration defines the maximum length of time (in seconds) that a call remains connected before the call is automatically disconnected. The default is **180** seconds. A call limit of three minutes may be too short for most configurations. Therefore, you may want to increase this default value.

Billing Options can be used to track the cost of Inbound and/or Outbound calls on any of the three interfaces (FXO, FXS, or E&M). The amount to be charged in cents is entered in the **Charge ( ) Cents** box together with the associated time duration in the **Per ( ) Seconds** box. While a given call is active, the accumulated charges can then be viewed on the **Call Progress** dialog box. When the call ends, the charges are transferred to a Log File that can be viewed by highlighting the call event in the **Log Entries** dialog box and selecting **Details**.

Dynamic Jitter Buffer defines a minimum and a maximum jitter value for voice communications. When receiving voice packets from a remote MultiVOIP 400, it is possible to experience varying delays between packets due to traffic conditions on the network. This is called Jitter. To compensate for this effect, the MultiVOIP 400 uses a Dynamic Jitter Buffer. The Jitter Buffer enables the MultiVOIP 400 to wait for delayed voice packets by automatically adjusting the length of the Jitter Buffer between configurable minimum and maximum values. An Optimization Factor adjustment controls how quickly the length of the Jitter Buffer is increased when jitter increases on the network. The length of the jitter buffer directly effects the voice delay between MultiVOIP 400 gateways.

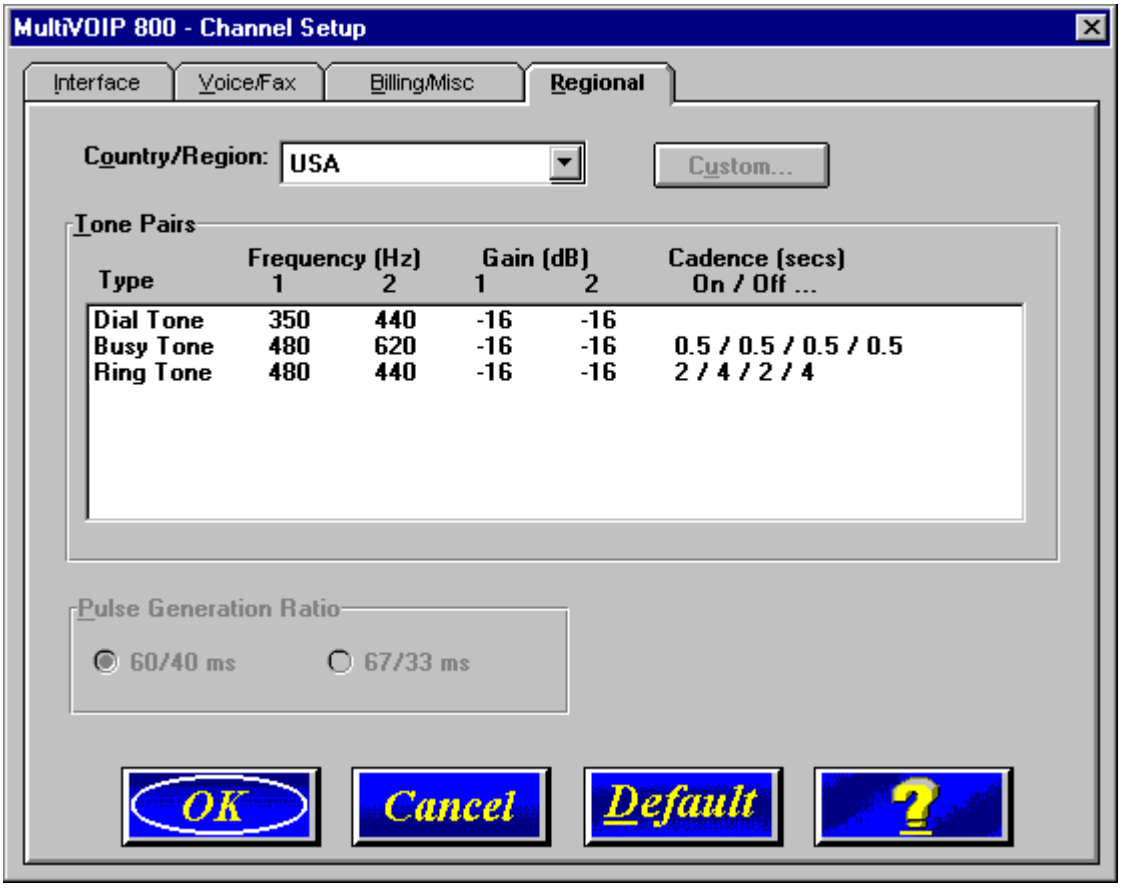
The default minimum dynamic jitter buffer of **150** milliseconds is the minimum delay that would be acceptable over a low jitter network. The default maximum dynamic jitter buffer of **300** milliseconds is the maximum delay tolerable over a high jitter network.

The **Optimization Factor** determines how quickly the length of the Dynamic Jitter Buffer is changed based on actual jitter encountered on the network. Selecting the minimum value of 0 means low voice delay is desired, but increases the possibility of jitter-induced voice quality problems. Selecting the maximum value of 12 means highest voice quality under jitter conditions is desired at the cost of increased voice delay.

The Optimization Factor can be configured in the range of 0 to 12 with a default setting of **7**.

Regional

The Regional tab controls the voice communications for the country or region in which the MultiVOIP 400 is being used.

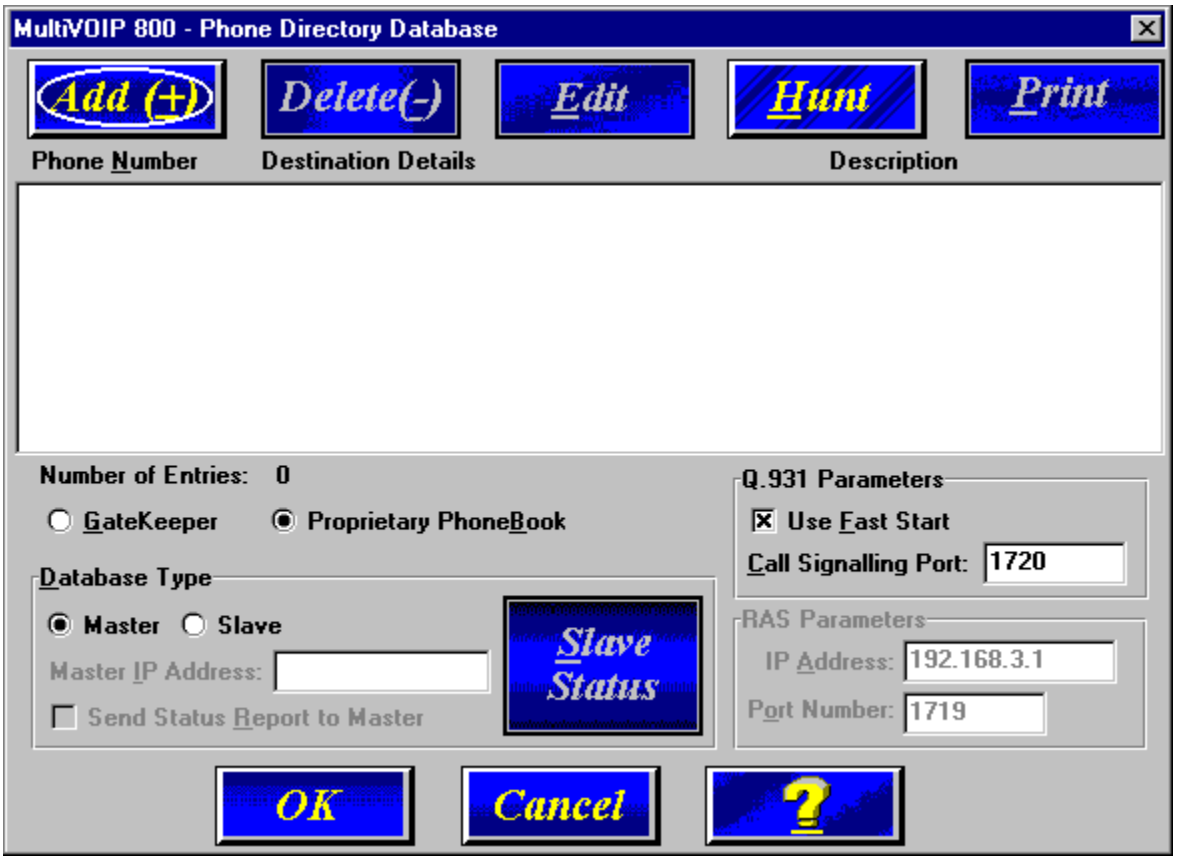


From the **Country/Region** list you can select the country or region for which you are configuring the MultiVOIP 400. The **Tone Pairs** group always displays the tones used in the country or region currently selected. In addition to Australia, Central America, Chile, Europe, France, Japan, UK, and USA, there is a Custom selection (with defaults identical to USA) that will make the **Custom** button active. Clicking **Custom** enables you to edit the Tone Pairs and establish custom sets of tone pairs for Dial Tone, Ring, and Busy on a **Custom Tone Pair Settings** dialog box.

The **Pulse Generation Ratio** group contains two ratios: the 60/40 is for the USA, and the 67/33 ratio is for international applications.

Changing the Phone Directory Database

The MultiVOIP provides two phone directory database architectures: the proprietary database and a database using an H.323 protocol gatekeeper that provides a centralized call control center. The proprietary database is based on a master and slave relationship in which the master VOIP maintains the phone directory database and distributes it to its slave VOIPs. The centralized call center establishes call control, call routing, address translation from LAN aliases to IP addresses, and bandwidth management. The H.323 protocol allows other third party gateways and endpoints that support the H.323 protocol standards to participate in the VOIP network (for example, Microsoft NetMeeting®).



The database displays the phone numbers in numerical order with destination details (IP Address or H323 ID), channel assignment, and a brief description of the entry (for example, New York Office 1). The method for changing the phone directory database is dependent on whether the **Gatekeeper** option or the **Proprietary Phonebook** option is enabled.

If the **GateKeeper** option is enabled, the **RAS Parameters** group is enabled and the IP address of the GateKeeper needs to be entered in the IP Address window. The Port Number is the port on the GateKeeper which it communicates with its endpoints. The **Q.931 Parameters** group is enabled in both the GateKeeper and Proprietary Phone Book Database architectures. The **Use Fast Start** option is used when the VOIP network supports Fast Start capability.

In the GateKeeper phone directory database, the phone directory database is developed through the **Add/Edit Phone Entry** dialog box. The **Add/Edit Phone Entry** dialog box defines the Station Information, phone number and voice channel of the unit, and station identification, H323 ID which defines the LAN alias and the IP address of the local unit. In the GateKeeper phone directory database, only the phone entries of the local unit display.

If the **Proprietary PhoneBook** option is selected, the **Database Type** group becomes active. This

defines the master-slave relationship. If the database type is master, then the **Add**, **Delete**, **Edit**, **Hunt**, and **Print** buttons at the top of the database dialog box are active. This allows the master database to establish the phone directory. The **Slave Status** button also becomes active in which you can view the active status of all the slave units.

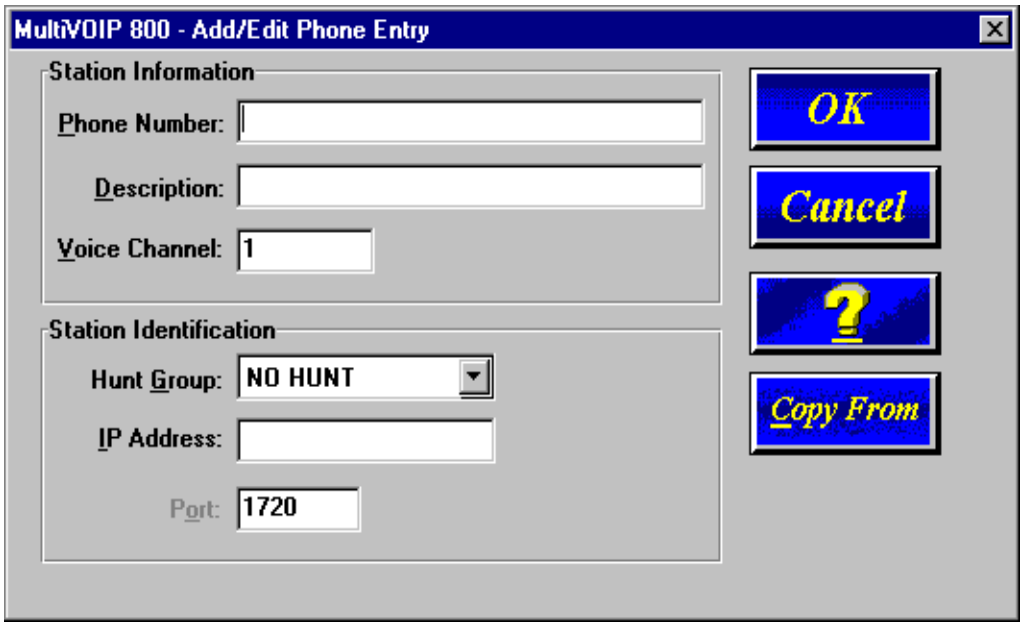
If **Master** is the selected database type, the phone directory database is developed through the **Add/ Edit Phone Entry** dialog box. The **Add/Edit Phone Entry** dialog box for the Proprietary PhoneBook defines the Station Information of the MultiVOIP, the phone number, description, the voice channel of the unit, and the station identification, which includes whether a hunt group is employed and IP address of the unit. This information is presented in the **Phone Directory Database** dialog box.

If the database type is set to slave, then the IP address of the master MultiVOIP needs to be entered in the Master IP Address window, the Send Status Report to Master option can be enabled, and all the buttons at the top of the directory database dialog box become inactive, except for the Print button.

Proprietary Phone Directory Database

In the **Proprietary Phone Directory Database**, you can add, delete, or edit any entry in the phone directory database and you can set up Hunt groups that locate another phone number if the called number is busy. You can print the phone directory database so that you have a hardcopy of the phone directory.

To add an entry to the Phone Directory database, click **Add** and the Add/Edit Phone Entry dialog box is displayed.



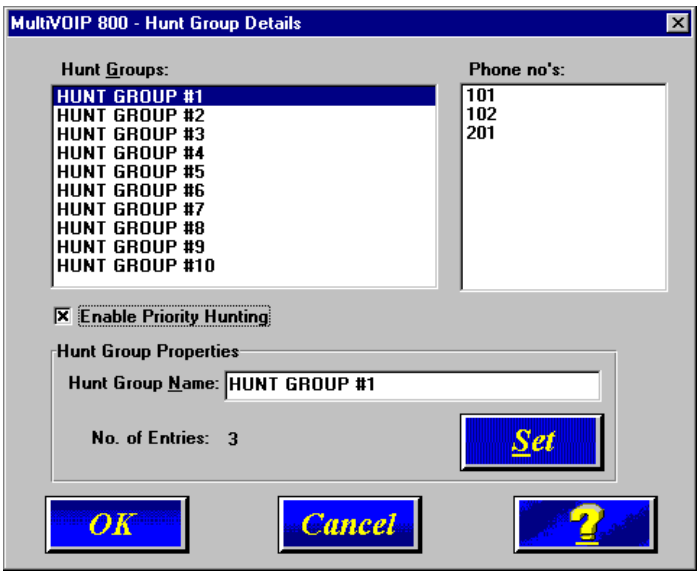
The **Add/Edit Phone Entry** dialog box contains two groups of information: the Station Information which contains the phone number, an optional description window, and the voice channel number. The **Station Identification** group contains the Hunt Group listing and the IP Address for the IP Address of the MultiVOIP assigned the phone number. The Port number is not used in the proprietary phone book. Click **Copy From** in the **Add/Edit Phone Entry** dialog box to add additional phone entries.

The **Station Information** identifies the calling unit by the phone number, a description if you choose, and voice channel of the unit doing the calling. The Phone Number does not have to be a conventional telephone number such as 555-0155. It can, for example, be a three-digit number such as 101. The Description window is like your name in your local telephone book listing. It identifies the calling party. For example, New York Office 1. The voice channel window defines the voice

channel associated with the telephone.

The **Station Identification** group enables you to assign the entry to a Hunt Group, provide the IP Address of the MultiVOIP being assigned the phone number, and accept the H.323 industry standard Port number. A Hunt Group is a series of telephone lines organized in such a way that if the first line is busy the next line is hunted and so on until a free line is found. It is a set of links which provides a common resource and which is assigned a single hunt group designation. A user requesting that designation may then be connected to any member of the Hunt Group.

You can view the details of the current Hunt Group (for example, HUNT GROUP #1) configuration by clicking **Hunt** in the **Phone Directory Database** dialog box. The current Phone numbers for HUNT GROUP #1 display.



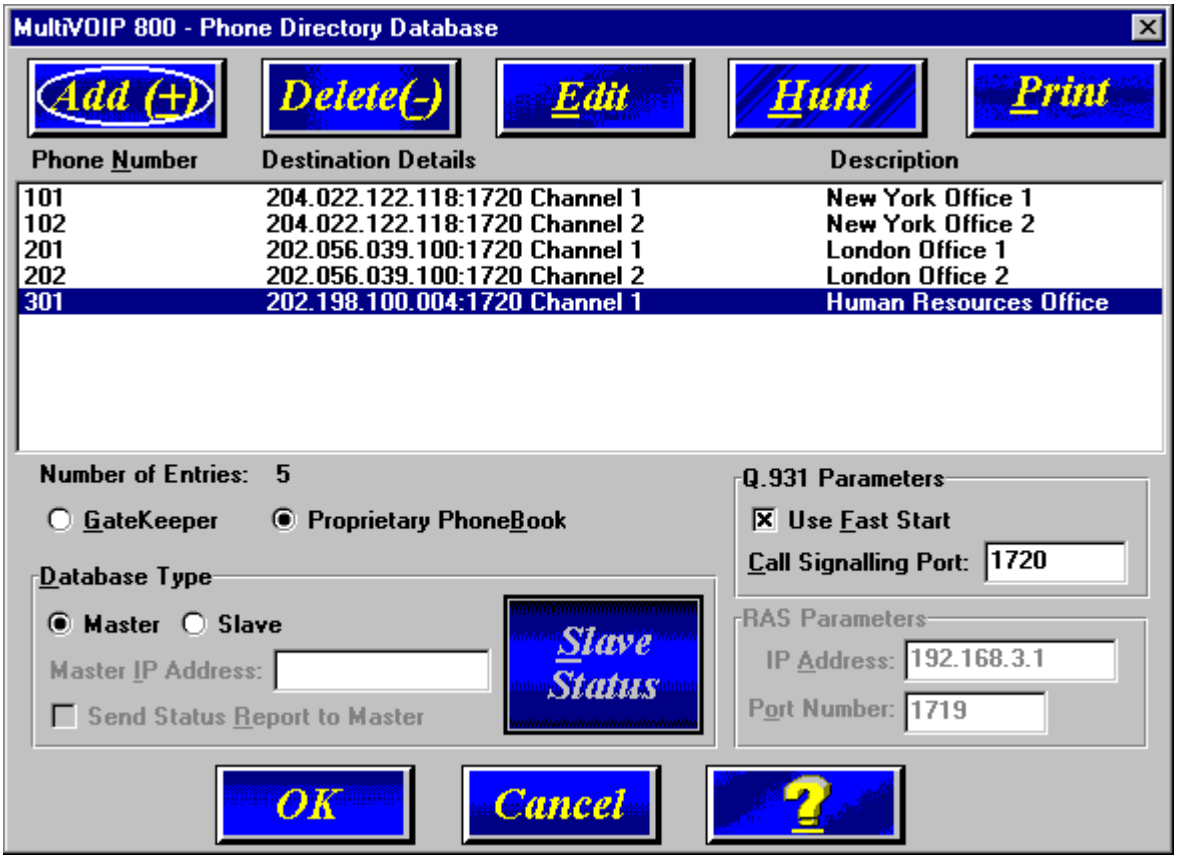
Select the Hunt Group you want to view. The **Phone no's** list displays the telephone numbers associated with that Hunt Group and the No. of Entries displays the running total of entries.

If the **Enable Priority Hunting** check box is selected, the number you dialed is given priority over other numbers in the same hunt group. If this check box is cleared, your call could be directed to any number in the hunt group.

Note: You can change the name of the Hunt Group by clicking on the entry that you want to change, editing the change in the Hunt Group name window, and then clicking **Set**.

Click **Slave Status** in the **Phone Directory Database** dialog box to view the status of all the slave units in your VOIP network. The **Send Status Report to Master** check box must be selected on the slave. The phone number of each slave displays with its IP Address, current line status, and the description of the phone number.

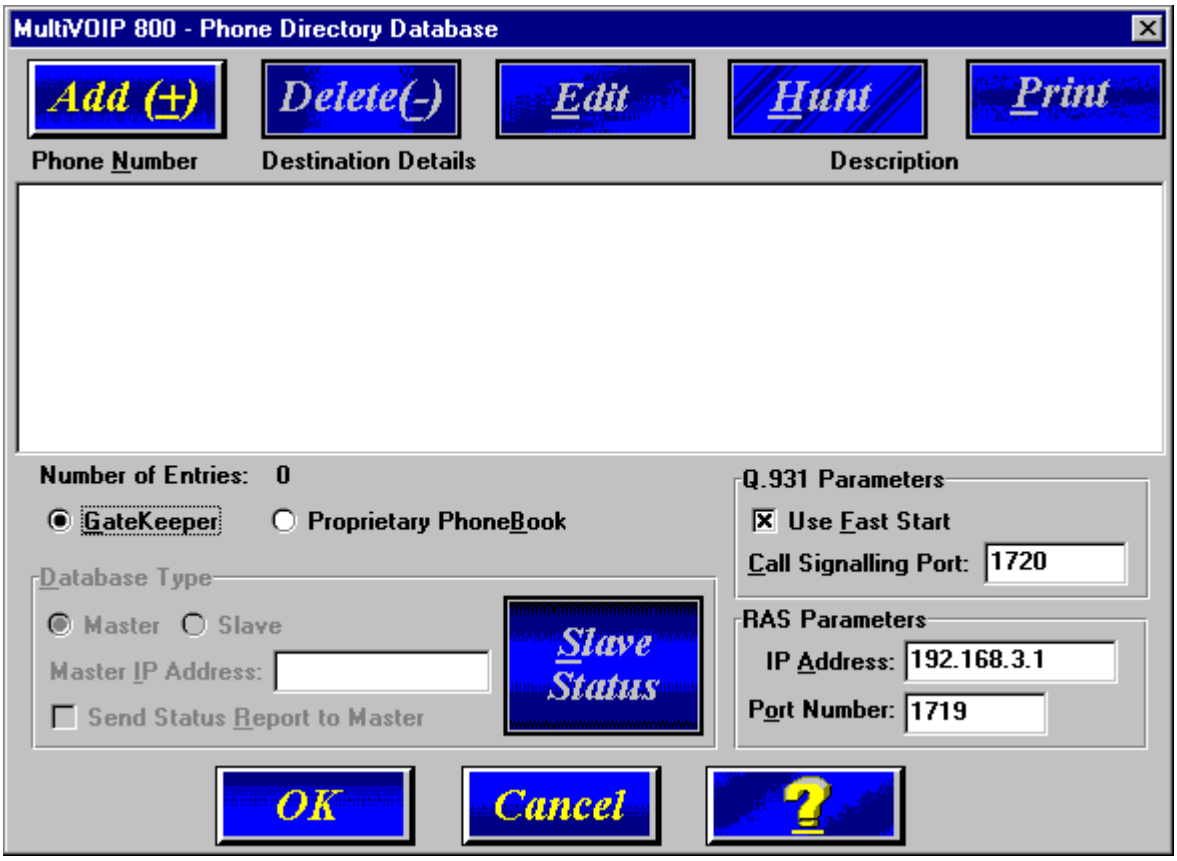
An example of a Proprietary Phone Directory database is shown below with the master MultiVOIP assigned phone numbers 101 and 102 in the New York Office and a slave unit with phone numbers 201 and 202 in the London Office. A third slave unit has phone number 301.



This Phone Directory database displays each slave unit in the VOIP network serviced by the master VOIP. When a slave unit comes online, the directory database is downloaded to the unit. So each VOIP in the network can see and call any other VOIP in the network. Only the master VOIP can change the phone number entries.

Gatekeeper Phone Directory Database

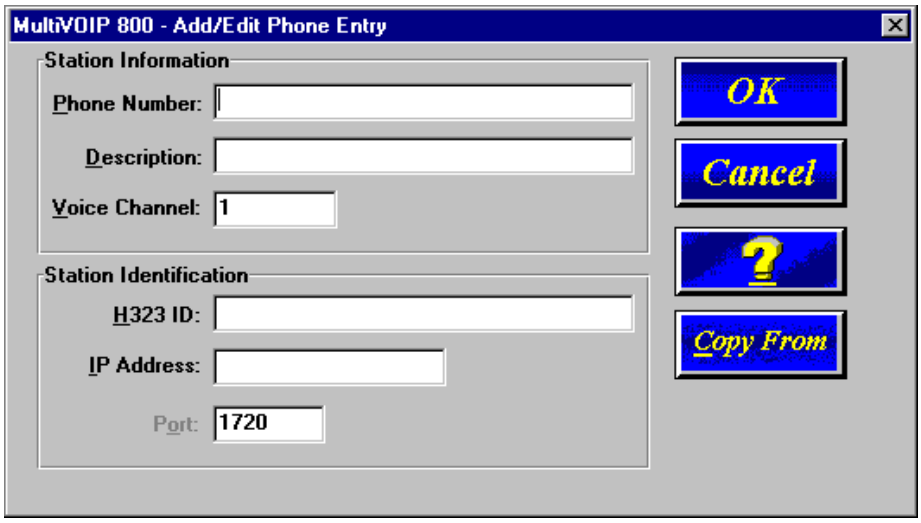
With the **Gatekeeper Phone Directory Database,** the Gatekeeper acts as the central point for all calls within its zone and provides call control services to registered endpoints. The Gatekeeper performs address translation from LAN aliases to IP addresses and provides bandwidth management where the network manager has specified a threshold for the number of simultaneous calls on the LAN.



When the **Gatekeeper** option is selected, the **RAS Parameters** group is enabled with the IP address of the Gatekeeper displayed in the **IP Address** box. The Port Number is the port of the endpoint communicating with the Gatekeeper. If this number is changed, it should only be changed with consultation with Gatekeeper administrator. The port numbers have to be in pairs and controlled by the Gatekeeper.

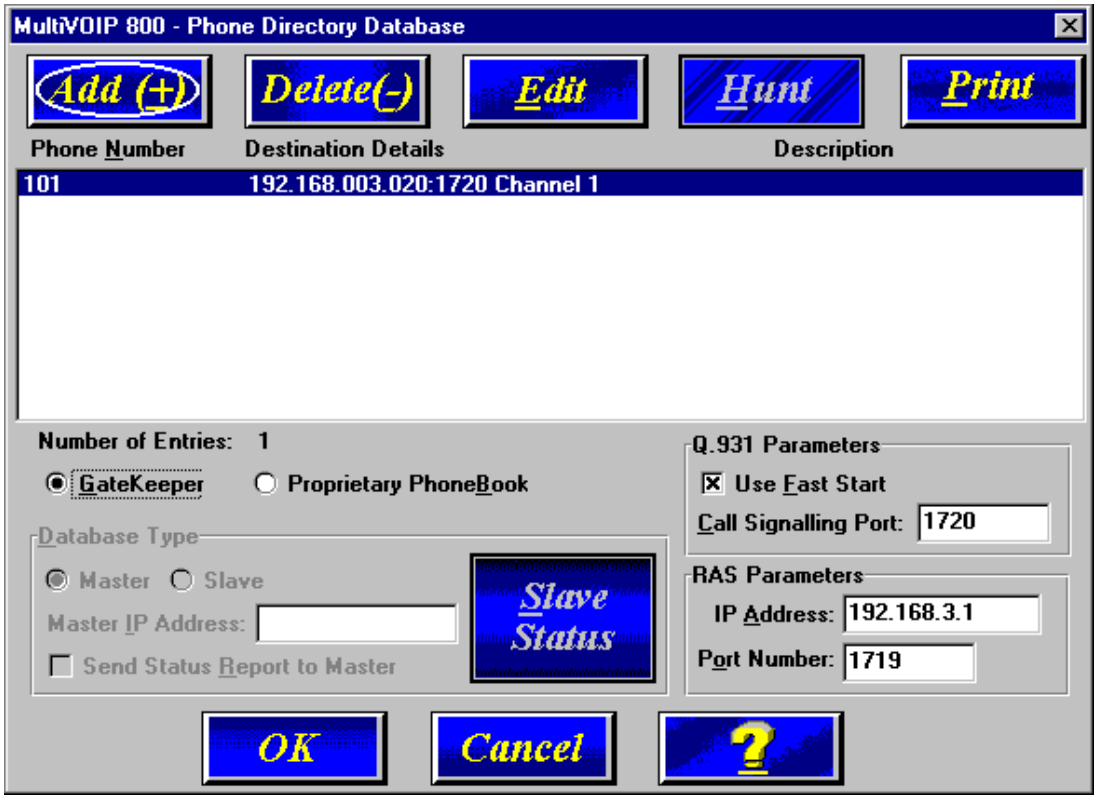
If the H.323 Gatekeeper network supports Q.931 Fast Start servicing, the **Use Fast Start** check box should be selected for all endpoints. The Call Signalling Port of 1720 is the port on the MultiVOIP unit supporting the Q.931 parameters.

The Phone Directory Database in a H.323 Gatekeeper network only displays the station information and station identification of the local unit. The station information and identification have to be established in conjunction with the Gatekeeper administrator so that the identification of the endpoint is the same. To add an entry to the Phone Directory Database, click **Add**. The **Add/Edit Phone Entry** dialog box displays.



The **Add/Edit Phone Entry** dialog box contains two groups of information. The **Station Information** group contains the phone number, the description window (which can be left blank), and the voice channel number. The **Station Identification** group contains the **H.323 ID** box and the **IP Address** box for the IP Address of the MultiVOIP assigned the phone number. The **Port** box contains the number of the MultiVOIP unit communicating with the Gatekeeper. This port number has to match the port number pair used by the Gatekeeper. If the port number on either end is changed, communication between the endpoint and the Gatekeeper is lost. Click **Copy From** in the **Add/Edit Phone Entry** dialog box to add additional phone entries.

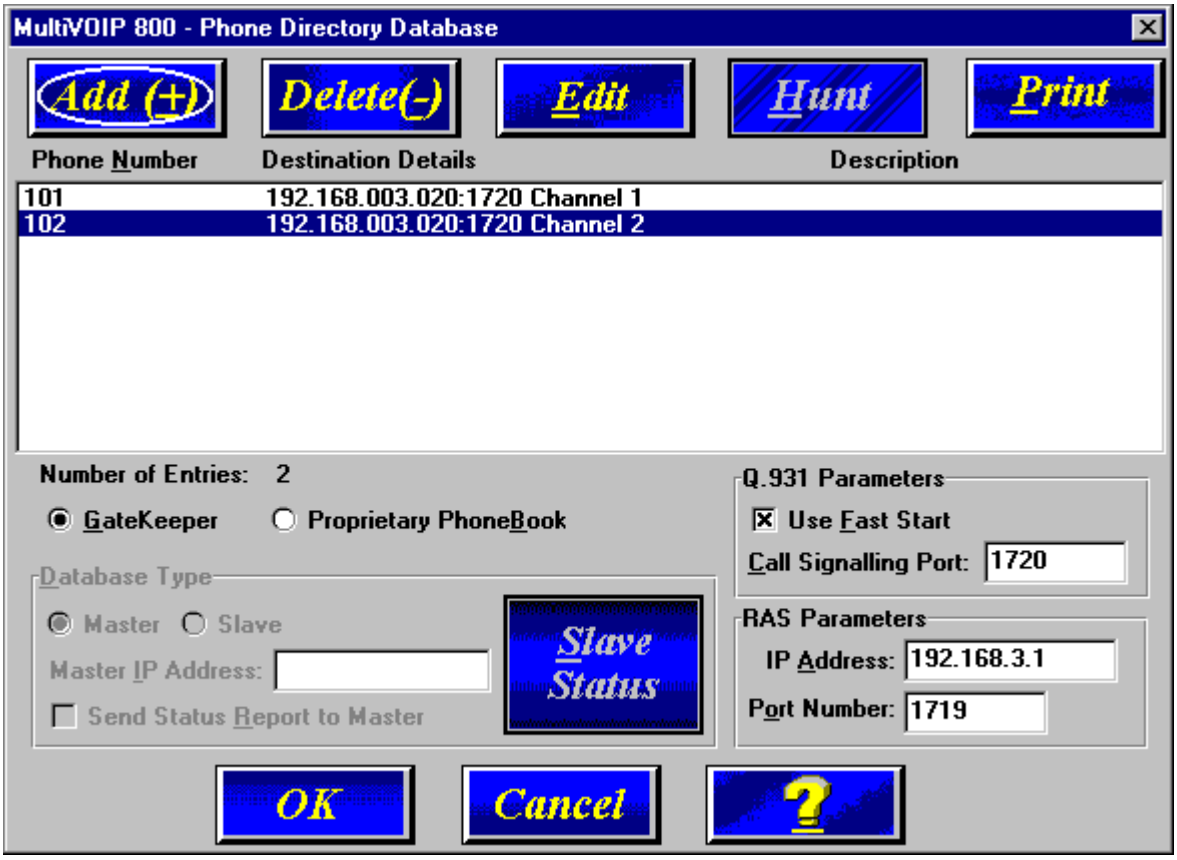
The **Station Information** group identifies the calling unit by the phone number, a description if you choose, and voice channel of the unit doing the calling. The phone number does not have to be a conventional telephone number such as 555-0155. It can, for example, be a three-digit number such as 101. The **Description** box and the **H323 ID** box may contain the same information. It is recommended that the **Description** box be left blank and the identifying information be entered in the **H323 ID** box. The voice channel window defines the voice channel associated with the telephone.



The **Station Identification** group allows you to define the MultiVOIP unit by establishing the H323 ID which can be the same as, for example, your name in your local telephone book listing. The Gatekeeper associates the H323 ID with the address of the local unit in the IP Address window. The Port number 1720 is the port of the MultiVOIP communicating with the Gatekeeper.

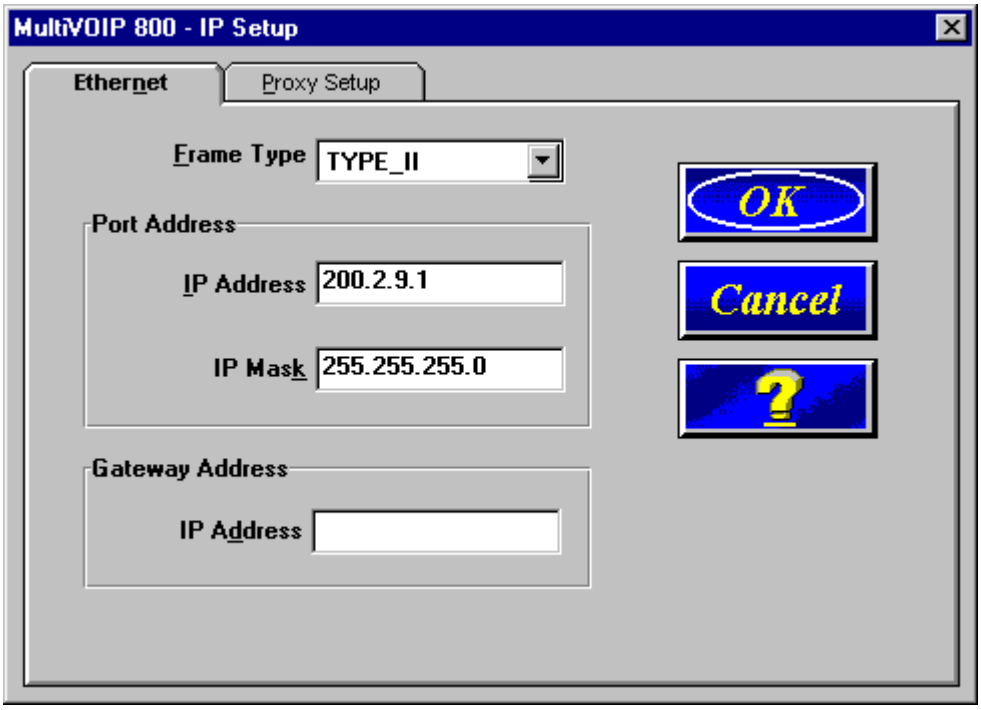
The Station Information and Identification of the MultiVOIP unit have to be identical the same type of information used by the Gatekeeper in order for the MultiVOIP unit to be registered with the Gatekeeper. The Gatekeeper can allow an open registration or a secure registration in which the endpoints are pre-defined by the Gatekeeper. The registration method is determined by the Gatekeeper administrator and will require communication with each endpoint in order to develop the H.323 compatiable network.

The Phone Directory Database for the local unit contains the local phone numbers, destination details of the IP address, port number, and channel number of the local unit.



Changing IP Parameters

The **IP Setup** dialog box establishes the IP addressing for the local Ethernet LAN and defines the Internet gateway address. The **IP Setup** dialog box is accessed by clicking the **IP** on the **MultiVOIP 800 Setup** menu.



With **IP Setup** dialog box displayed you can change the status of differential services, the Ethernet Frame Type, the IP address and IP Mask of your H.323 endpoint, and the Gateway Address of the IP address of the device connected to the Internet.

Selecting the **Enable Diffserv** check box enables Differentiated Services. The check box is cleared by default. Differentiated Services provides priority to voice packets so that they are not delayed whenever large data files need to be downloaded. If any routers in the VOIP network support this feature, you should enable this function.

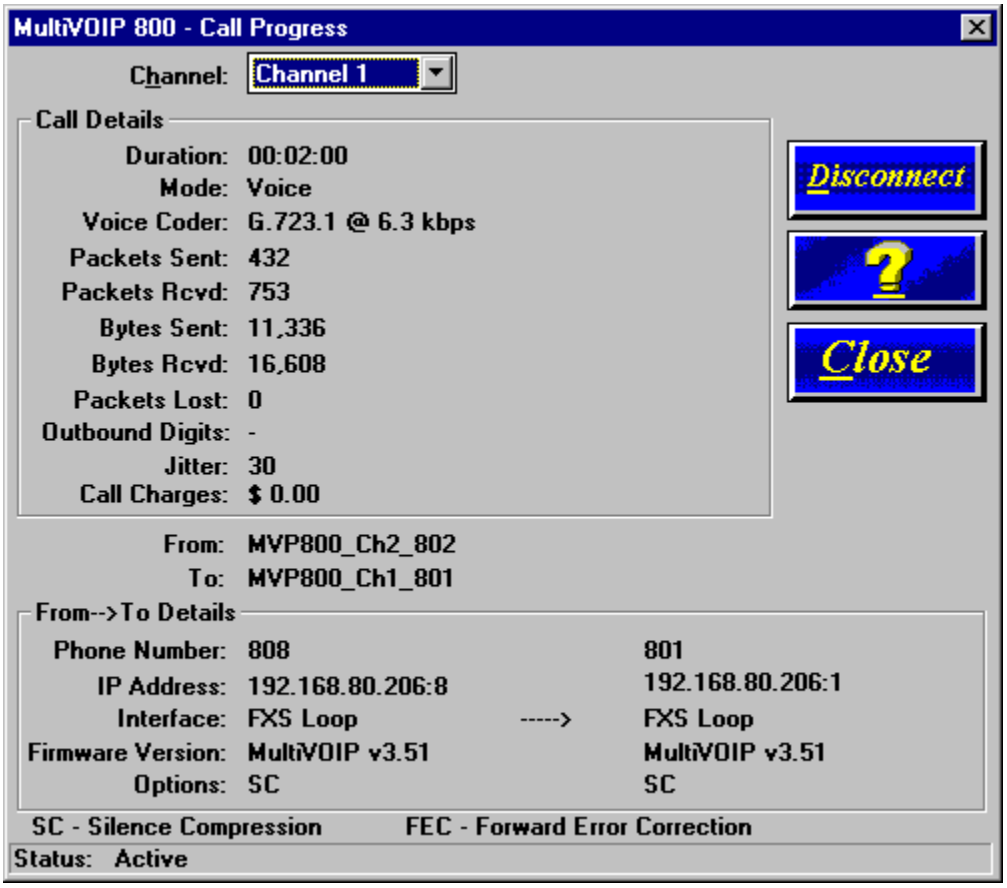
The **Frame Type** list enables you to change the Ethernet Frame Type so that it matches your IP network. If the current entry does not match your IP network, select a new Frame Type from the list. Frame Type choices are **TYPE_II** and **SNAP**.

The **Port Address** group enables you to change the unique **IP Address** and **IP Mask** of the local LAN.

The **Gateway Address** group enables you to change the gateway **IP Address** of the device connected to the Internet/Intranet.

Viewing Call Progress

The **Call Progress** dialog box displays the status of a call in progress. This dialog box is accessed from the MultiVOIP 800 Setup menu by clicking **Call Progress**.



The ratio of Packets Lost versus Packets Received provides a general indication of the integrity of the Internet connection. To reduce the frequency of lost packets, select a low-bit-rate coder, such as, G.723 or Netcoder. Selecting the **Forward Error Correction** check box on the **Voice/Fax** tab in the **Channel Setup** dialog box enables the MultiVOIP 400 to recover many of the lost packets.

The Jitter value (measured in milliseconds) indicates the mean deviation of the difference in packet spacing at the receiver compared to the sender for a pair of packets.

Applications Setup

Click **Others** on the Setup menu to open the **Applications Setup** dialog box. You can enable the SNMP Agent (the default is disabled), set up all the necessary parameters, enable or disable various remote configuration methods such as TFTP (Trivial File Transfer Protocol) Server, Web Server, Dumb Terminal Management, and Telnet Server, and assign a Password to the MultiVOIP 400 for Internet security. These remote configuration applications enable remote viewing and changing of the MultiVOIP 400 configuration, or updating firmware, from anywhere on the connected internetwork.

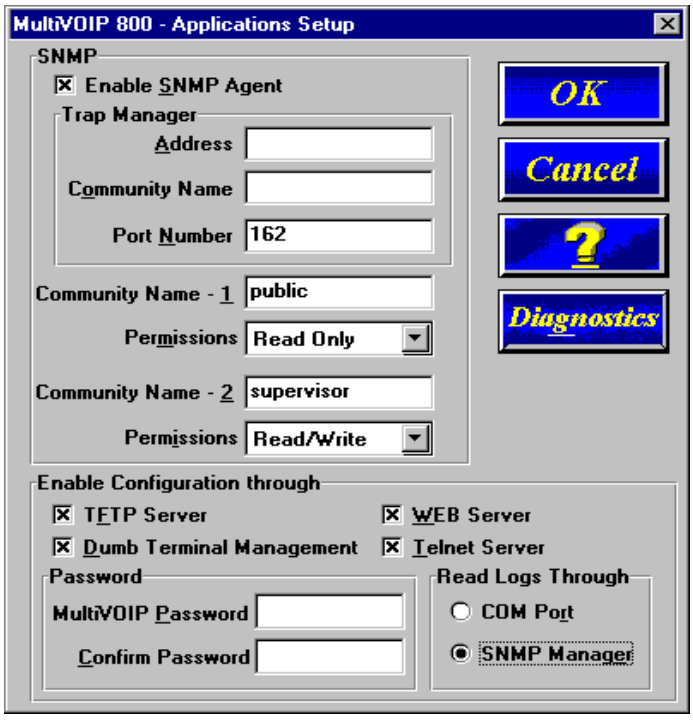
Verify that the desired applications are selected. All applications in the **Enable Configuration through** group are selected by default. To disable a given application, click to clear the check box and disable support.

SNMP related operations can be performed only when the **Enable SNMP Agent** check box is selected on this dialog box. Enter the IP address of the system (such as SNMP Manager) that will receive the Traps from the MultiVOIP 400 in the **IP Address** box. The **Community Name** of the SNMP Manager receiving the Traps can be a maximum of 19 characters and is case-sensitive. The default **Port Number** of the SNMP Manager receiving the Traps is 162. The MultiVOIP 400 currently supports a maximum of two community users at a time. They can be assigned either Read/Write or Read Only rights.

If you have SNMP client software and the **Enable SNMP Agent** check box is selected, the **SNMP Manager** option must be selected. This enables you to read logs through the SNMP Manager instead of the COM port.

The **Password** group enables you to enter a password, up to 13 alphanumeric characters, to be used for Internet Security. Once the password is entered in the **MultiVOIP Password** box and confirmed in the **Confirm Password** box, remote users must enter the password before gaining access to the MultiVOIP 400 for configuration purposes.

Note: If you forget your password, contact Multi-Tech Technical Support for instructions.



For more information on using these applications, refer to the Help or to Chapter 5 of this manual, Remote Configuration and Management.

MultiVOIP™

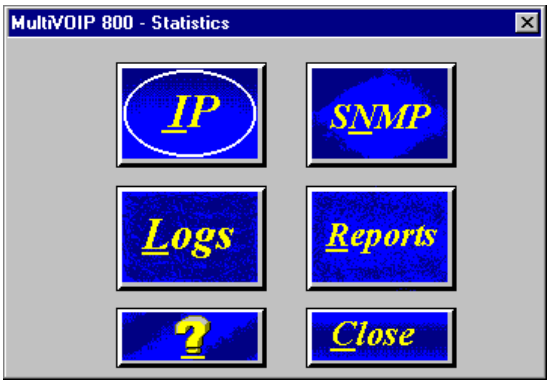
Voice/Fax over IP Networks

Chapter 5 - Remote Configuration and Management

MultiTech® Systems

Viewing Statistics

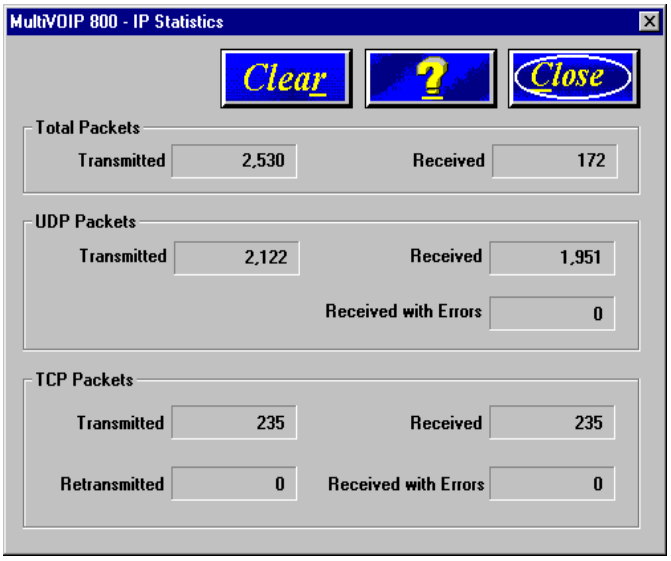
The **Statistics** dialog box enables you to view statistics for major events of the MultiVOIP 400 operation. This dialog box is accessed by clicking **Statistics** on the **MultiVOIP 800 Setup** menu.



Statistics can be a helpful troubleshooting tool. For example, viewing the Voice Channel statistics you can see the attempted and completed calls, call duration, average call length, bytes/packets sent and received, and so on.

IP Statistics

IP is a connection-less network protocol residing in the network layer of a conventional OSI layered model. Refer to Appendix A for more information on this model. Depending on what is going on at the application layer, IP will typically use one of two transport layer protocols: User Datagram Protocol (UDP), a connection-less transport layer protocol used with TFTP or SNMP, and Transport Control Protocol (TCP), a connection-oriented transport layer protocol used with FTP, Telnet, and SNMP.



UDP makes use of the port concept and has no measures for flow control, reliability, or error recovery. It is used when the full services of TCP are not required, and the reliability measures must be assumed by another layer.

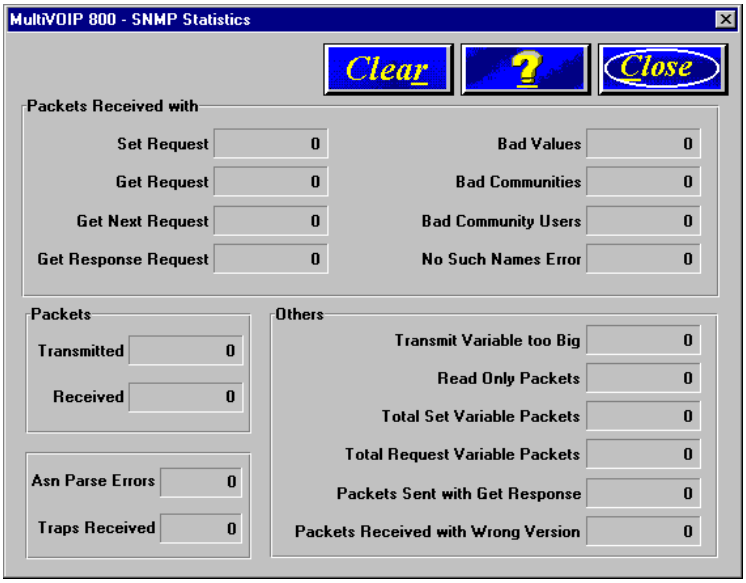
TCP works well in environments where the reliability measures are not assumed by other layers. It is connection-oriented and has a full range of services.

For the most part, these statistics are informational. Their use as a troubleshooting tool will be contingent on the applications running in the upper layers. For example, if you were having problems

connecting to the MultiVOIP 400's web server, you would look under the TCP section to see if any connections are being established. If not, that may indicate the web server is not enabled. Or, if you were having problems establishing a remote connection through TFTP, you could look in the UDP section to see if any packets are being received. If not, you may need to review your network addressing.

SNMP Statistics

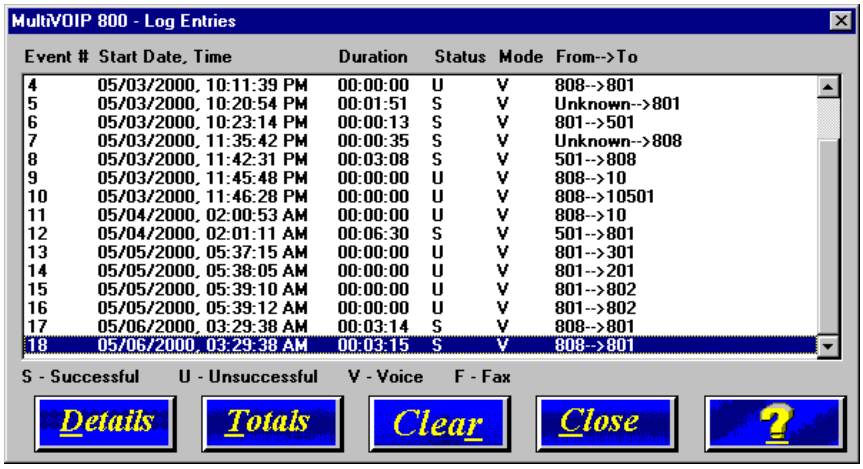
The **SNMP Statistics** dialog box provides statistical information on Simple Network Management Protocol (SNMP).



SNMP is an application layer protocol that facilitates the exchange of management information between network devices. There are three key components in SNMP: (1) the devices being managed, (2) the agents, and (3) the network management systems. The managed device is the network device, like a router. The agent is the software module residing in the managed device pertaining to network management. The network management system runs the SNMP application that controls the managed devices and monitors their status. Four primary operations (Set, Get, Get Next, and Trap) are performed using SNMP.

Viewing Logs

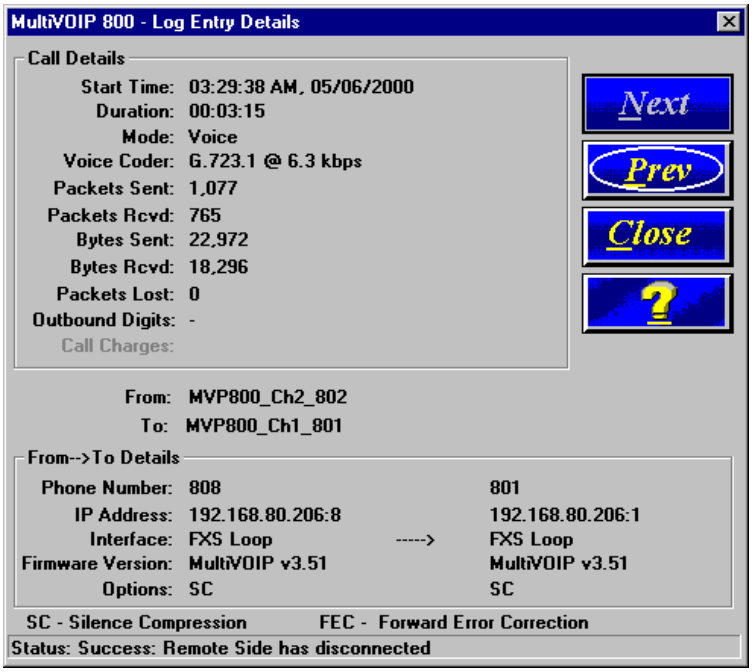
The **Log Entries** dialog box displays a chronological history of all calls into and out of this unit. Click **Logs** in the **Statistics** dialog box to open this dialog box.



The **Log Entries** dialog box displays each call as a sequentially numbered Event with the date, time, duration of the call, the status of the call (**S**uccessful or **U**nsuccessful), Mode (**V**oice or **F**ax), and the from and to numbers.

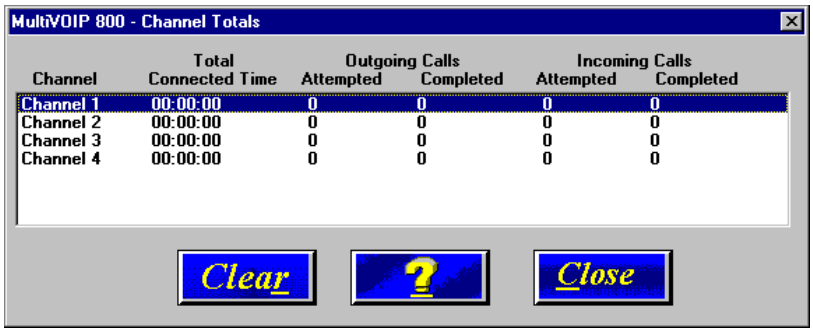
Viewing Log Entry Details

The **Log Entry Details** dialog box displays the status of a completed call. This dialog box displays the same details as the **Call Progress** dialog box after a call is completed.



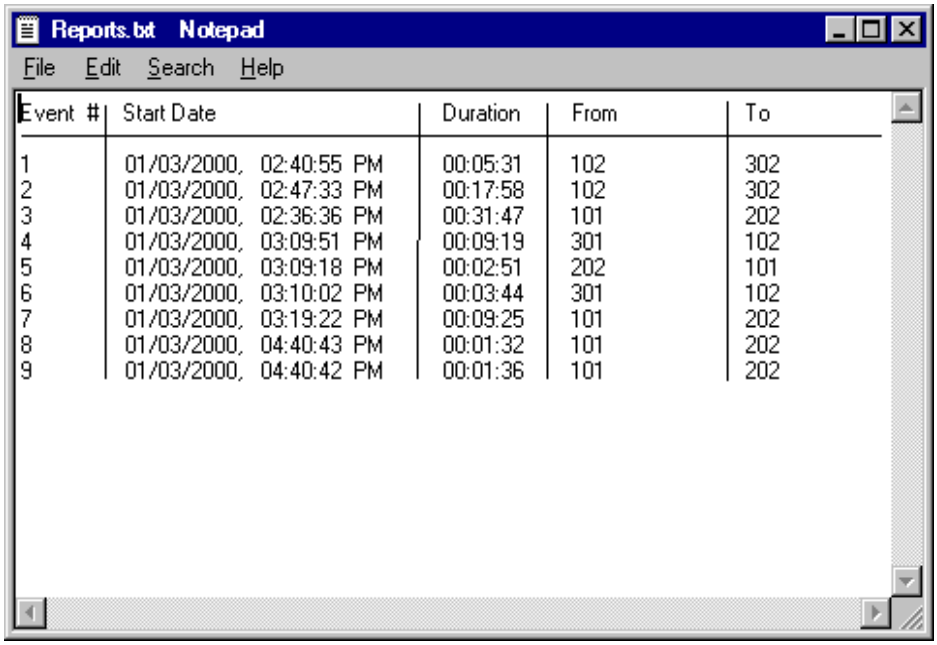
Viewing Channel Totals

The **Channel Totals** dialog box displays Outgoing and Incoming calls with their Attempted and Completed numbers for each channel on this MultiVOIP 400. The Total Connected Time for the channel also displays. This provides a listing of successful call completions on each channel of the unit.



Reports

A report of the contents of the **Log Entries** dialog box can be generated using the Windows Notepad accessory and then printed from your local PC. The report is generated by entering the To and From dates in the **Report Generation** dialog box and then clicking **Generate**. This function provides a hard copy of the **Log Entries** dialog box.



Upgrade Procedures

Whenever you upgrade your version of the MultiVOIP 400 software, you must first install the new software on your PC. Then, download the Firmware, the Factory Defaults, the Voice Coders, and/or the H.323 stack to upgrade the MultiVOIP 400 itself.

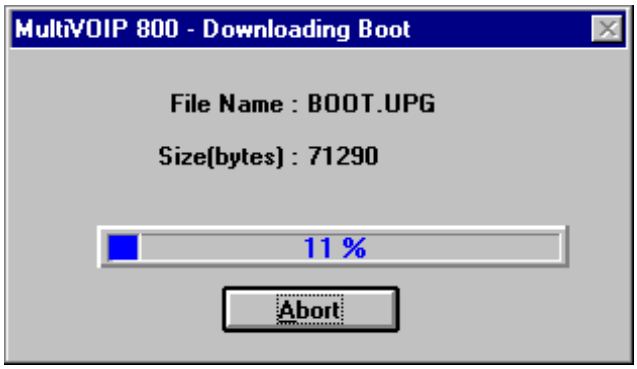
Before starting the upgrade process, view the current configuration and write down important data such as your IP address and voice channel configurations. These settings must be put back in place after upgrading the software. Multi-Tech also recommends that you click **Print** in the **Phone Directory Database** dialog box to print a copy of the phone directory contents.

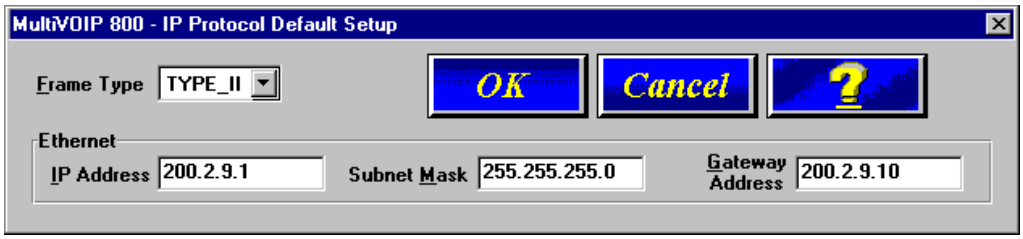
Three utility programs included in the MultiVOIP 400 software are to be used only after the unit has been repaired or upgraded. They are **Download Firmware**, **Download Voice Coders**, and **Download H.323 Stack**.

Download Firmware

If you have obtained a new firmware version from the Multi-Tech Web site, the Multi-Tech FTP site, or another source, do the following:

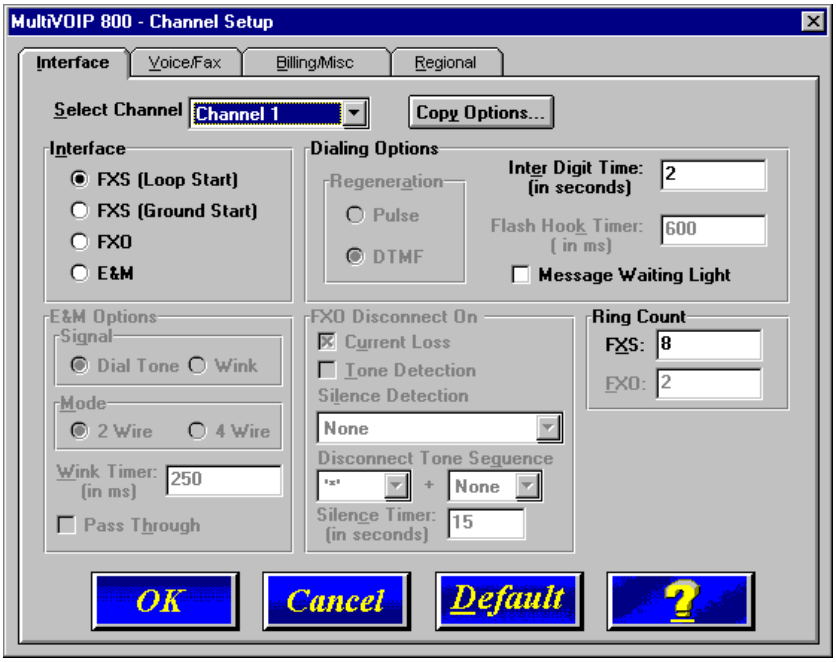
From your desktop, click **Start | Programs | MultiVOIP 800 v3.51| Upgrade Software**. The **Open** dialog box displays.



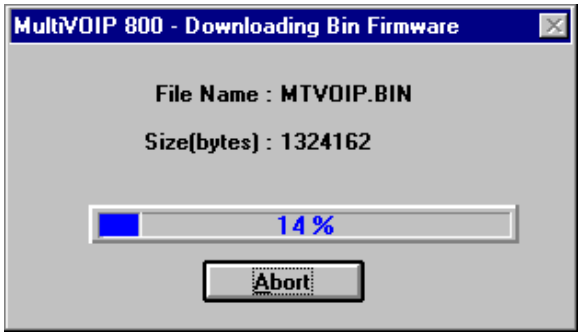


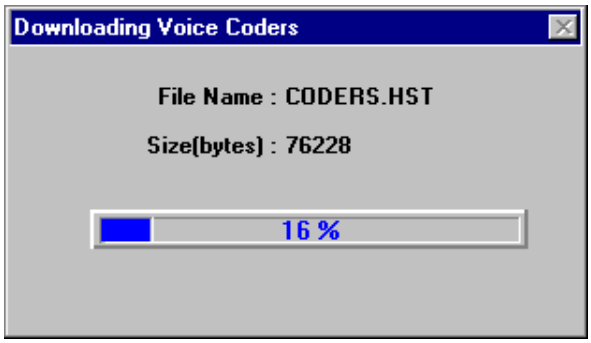
Enter your LAN IP Address, Subnet Mask, and Gateway Address in the **IP Protocol Default Setup** dialog box. Click **OK** when finished.

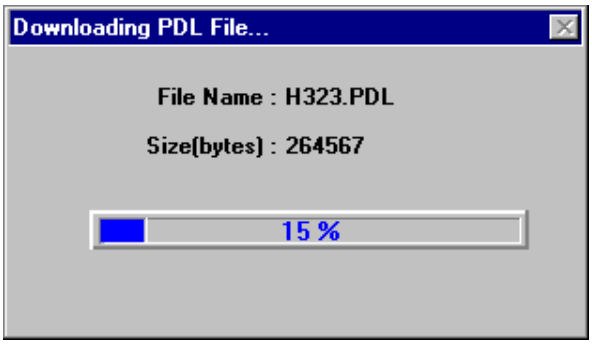
Enter your current configuration in the **Channel Setup** dialog box. Click **OK** when finished.



The MultiVOIP firmware, coders, and H.323 stack download, then the MultiVOIP reboots.







Manual Upgrade Procedure

Use this procedure to MANUALLY upgrade your previously-installed MultiVOIP400s from 3.01D or earlier to H.323 (version 3.51)

MVP400 version 3.51 MANUAL Installation Instructions

Follow this procedure ONLY if you want to MANUALLY upgrade your MultiVOIP.

Note: Steps 5-8 must be performed locally via the command port.

Note: The MVP 400/800 must have 8 MB of Ram to function properly in the H.323 mode. First, verify that two SIMMs are installed. Remove the front panel of the unit. If there are two SIMMs installed, you have 8 MB of RAM. If not and you are upgrading a MVP400, you will need to install a second 4 MB SIMM into the empty SIMM socket. The installed SIMM should be a 72-pin dram/simm/fast page mode, not EDO. The SIMM is available from Supplynet (P/N 94099400). Visit their Web site at www.thesupplynet.com or call 800-826-0279.

1. Run “MultiVOIP configuration” from your old version of MultiVOIP software and take note of the current settings. Your MultiVOIP will be reset to factory defaults during this upgrade.
2. Uninstall your old version of MultiVOIP software by selecting the “Uninstall MultiVOIP Configuration” option from the program group.
3. Install the H.323 compatible software from the MultiVOIP CD.
4. From the program group of the MultiVOIP software, run “Download H.323 Stack” and select the default file.
5. From the program group of the MultiVOIP software, run “Download Voice Coders” and select the default file.
6. From the program group of the MultiVOIP software, run “Download Factory Defaults”.
7. From the program group of the MultiVOIP software, run “Download Firmware”. Under “File Name”, enter “*.upg” and press ENTER. Select the file named boot.upg.
8. From the program group of the MultiVOIP software, run “Download Firmware” and select the file mtvoip.bin.
9. Run “MultiVOIP Configuration” and reconfigure your MultiVOIP. The upgrade is complete.

Introduction

This chapter provides procedures for viewing or changing the configuration of a remote unit. Two methods are provided to access a remote unit: the first is modem based and the second uses IP. Within the IP method, three applications can be used: 1) LAN-Based using TFTP (Trivial File Transfer Protocol), 2) Telnet as a client application, or 3) a standard web browser on the Internet.

Remote Configuration

Remote configuration requires the MultiVOIP 400 software to be loaded on the local PC. The local PC then controls the remote MultiVOIP 400 either via the modem connection or the LAN.

Modem-Based

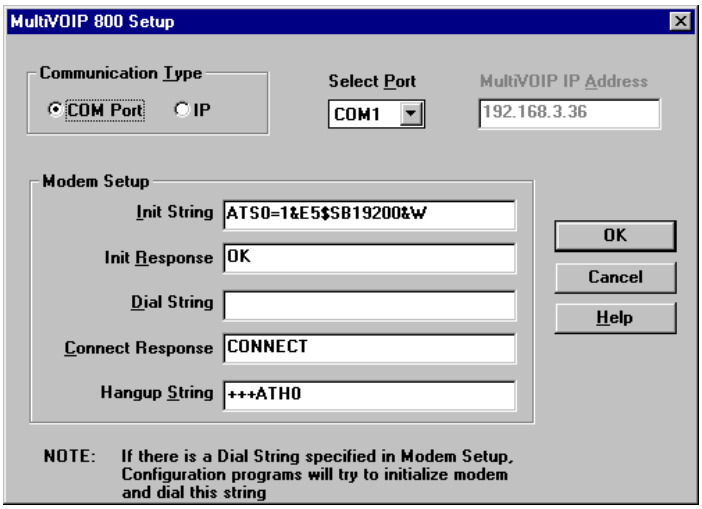
To remotely configure a MultiVOIP 400, a local PC needs to be connected to a dial-up line and the MultiVOIP software configured to call the remote MultiVOIP. The remote MultiVOIP needs to have a modem connected to a dial-up line and the Command Port. Once the connection to the remote unit is made, you can change the configuration as required. Once the configuration is changed, you can download the new configuration to the remote MultiVOIP. Refer to the Modem-Based Remote Configuration Procedure in this chapter to remotely configure a MultiVOIP 400.

1. At the remote site, remove the serial cable from the PC to the Command Port connector on the back panel of the MultiVOIP 400.
2. At the remote site, connect a special cable (Remote Configuration Cable) from the Command Port connector on the back panel of the MultiVOIP 400 to the RS-232 connector on the modem. The special cable is a serial cable with male connectors on both ends. Refer to Appendix B for cable details.

   Connect the modem to your local telephone line.

   Provide your telephone number to the person verifying your configuration.

   Configure the remote modem for 19200 baud and turn on Force DTR.
3. At the main site, connect your local PC to a modem that is connected to a dial-up line.
4. Install the MultiVOIP 400 software on the local PC. When installed, click **Start | Programs | MultiVOIP 800 v3.51| Configuration Port Setup**, or double-click on the **Configuration Port** icon in the MultiVOIP 400 program group.
5. The **MultiVOIP 800 Setup** dialog box displays.



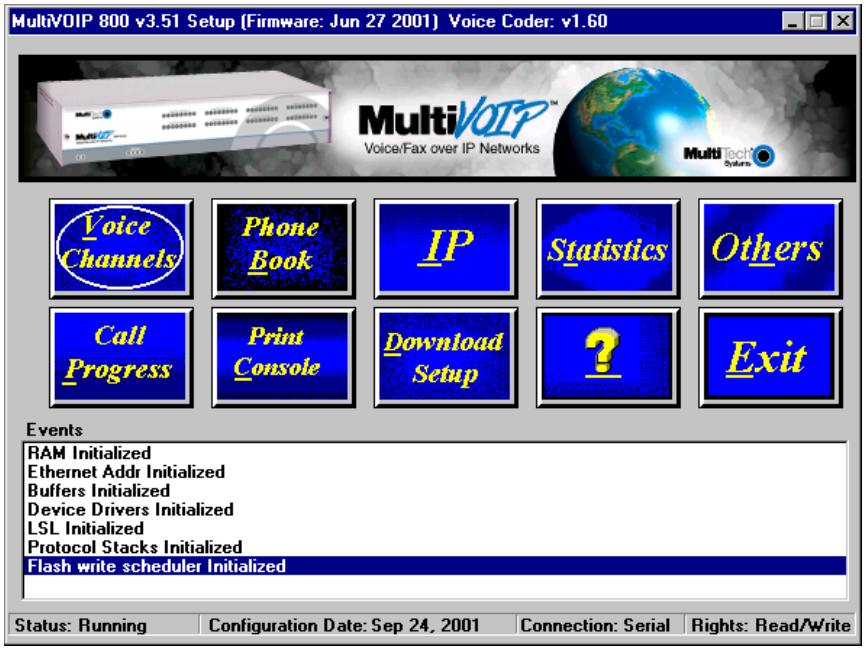
Verify that the **Communication Type** is set for **COM Port** and the **Select Port** box is set for the COM port of your local PC.

In the **Dial String** box, enter the AT command for dialing (ATDT) plus the phone number of the remote MultiVOIP 400.

If your Modem Initialization String, Initialization Response, or Connect Response values are different than the defaults in the dialog box, refer to your modem user documentation and change the default values to match your modem.

Click **OK** when you are satisfied with your selections.

6. Run the MultiVOIP 400 Configuration program. Click **Start | Programs | MultiVOIP 800 v3.51| MultiVOIP Configuration**, or double click on the **MultiVOIP Configuration** icon in the MultiVOIP 400 program group.
7. The **Dialing** dialog box displays while software is dialing the remote MultiVOIP 400.
8. The **Reading Setup** dialog box displays.
9. The **MultiVOIP 800 Setup** menu displays. This is the dialog box of the remote MultiVOIP 400. Refer to the Help for more information.



10. After you have changed the configuration of the remote MultiVOIP 400, click **Download Setup** to update the configuration. The remote MultiVOIP 400 will be brought down, the new configuration written to the unit, and the unit will reboot.
11. Click **Exit** when downloading is complete.
12. The **Hangup connection?** dialog box displays.

    Click **Yes** to disconnect the phone connection to the remote site.
13. If the same telephone number is not going to be used again in the immediate future, you may want to remove it from the **Port Setup** dialog box.

LAN-Based

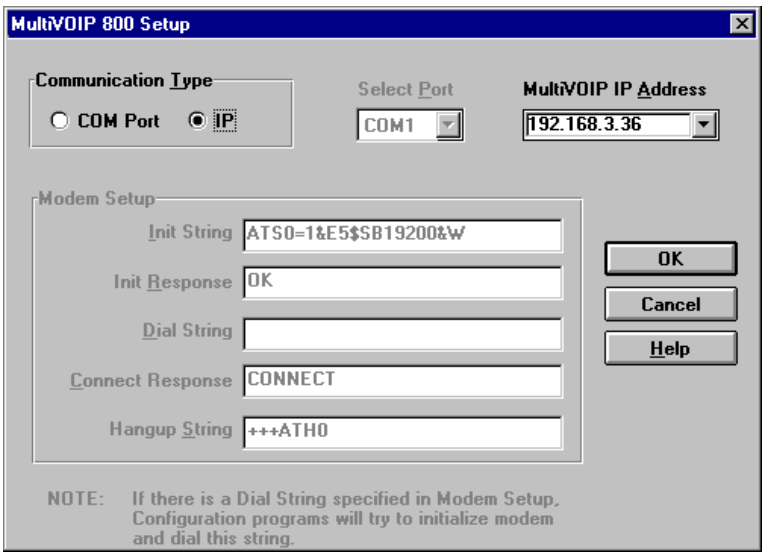
The LAN-based remote configuration requires a Windows Sockets compliant TCP/IP stack. TCP/IP protocol software must be installed and functional before the configuration program can be used.

1. You must assign an Internet (IP) address for the PC and for each node that will be managed by the configuration program. Refer to the protocol software documentation for instructions on how to set the IP addresses.

   Once you have completed this step, you should be able to use the protocol Ping command for the PC host name. You should also test the network interface configuration by Pinging another TCP/IP device that is connected to the network.

2. Install the MultiVOIP 400 software on the local PC. When installed, click **Start | Programs | MultiVOIP 800 v3.51| Configuration Port Setup**, or double-click the **Configuration Port Setup** icon in the **MultiVOIP 400** program group.

3. The **MultiVOIP Port Setup** dialog box displays.



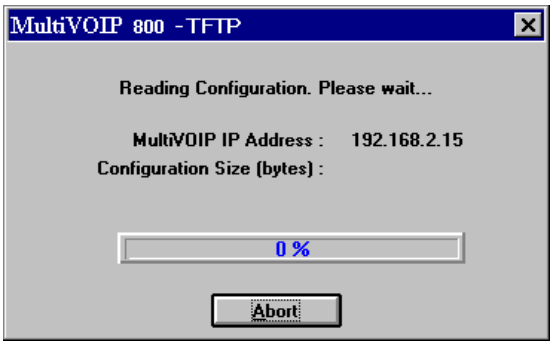
   Verify that **IP** is set for the Communication Type.

   In the **MultiVOIP** 400 **IP Address** box, enter the IP Address of the remote MultiVOIP 400.

4. Click **OK** when you are finished making selections.

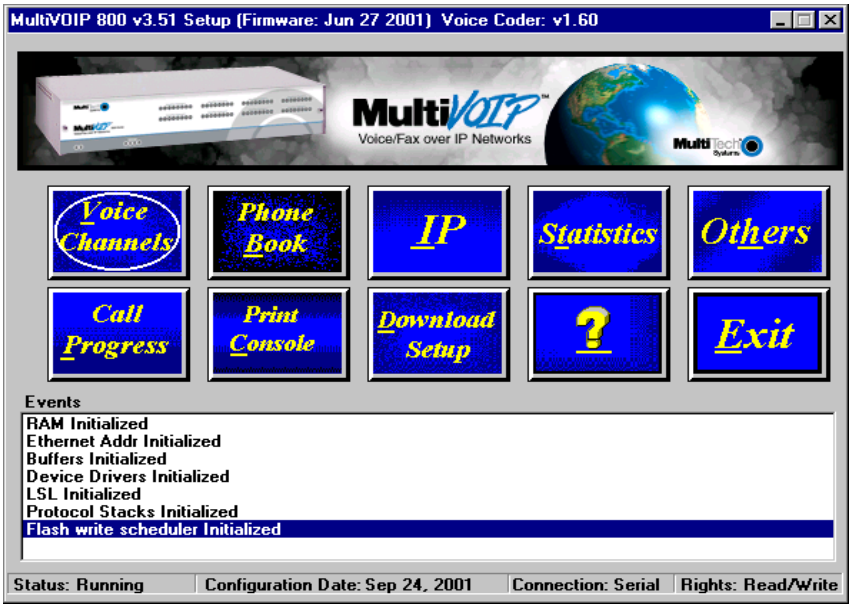
5. Run the MultiVOIP 400 Configuration program. Click **Start | Programs | MultiVOIP 800 v3.51| MultiVOIP Configuration**, or double-click the **MultiVOIP Configuration** icon in the **MultiVOIP 400** program group.

   The **Reading Setup** dialog box displays.



6. The **MultiVOIP 800 Setup** dialog box displays. This is the dialog box of the remote MultiVOIP

400. Refer to the Help for the more information.



7. After you have changed the configuration of the remote MultiVOIP 400, click **Download Setup** to update the configuration. The remote MultiVOIP 400 will be brought down, the new configuration written to the unit, and the unit will reboot.
8. Click **Exit** when downloading is complete.
9. Double-click the **MultiVOIP Configuration** icon in the **MultiVOIP 400** program group to verify that the MultiVOIP 400 is running.

Remote Management

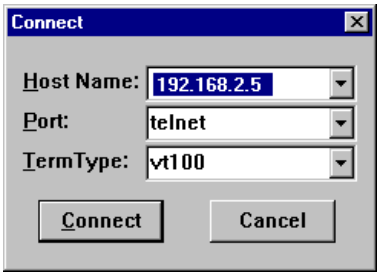
This section describes typical client applications that can be used to configure the MultiVOIP 400 remotely. It is important to note that although any subsequent changes to configuration can be made using these applications, the initial setup and configuration of the MultiVOIP 400 must be done on the local PC using the MultiVOIP 400 software provided with your unit.

Although establishing access to the MultiVOIP 400 varies between applications, the configuration functions mirror those of the MultiVOIP 400 software. For more information on MultiVOIP 400 software, refer to Chapter 4, MultiVOIP 400 Software.

Telnet

A typical Telnet client application is described next. The MultiVOIP 400 has a built-in Telnet Server that enables Telnet client PCs to access, and configure, the MultiVOIP 400. In addition, the MultiVOIP 400 can be remotely accessed and configured from anywhere on the Internet through its Web interface.

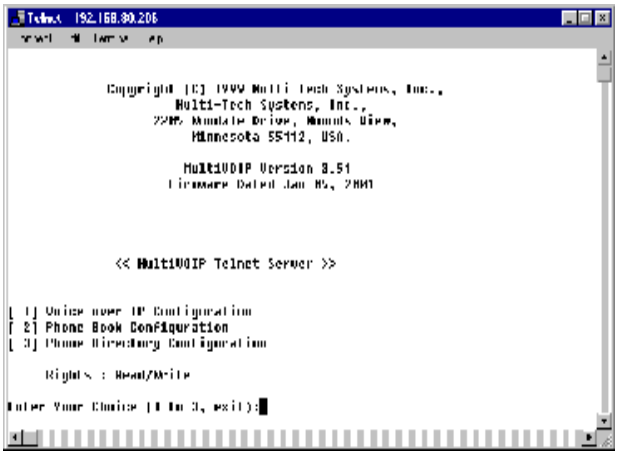
The TCP/IP stack has to be loaded before the Telnet client (a Windows application) will run. The Telnet Server option has to be selected from the **Applications Setup** dialog box using the **MultiVOIP 400 Configuration** icon. In your Windows 95/98/NT PC, click **Start** and then click **Run**. Type **Telnet** , then click **OK**. A blank Telnet window displays. Click **Connect** I **Remote System** and the **Connect** dialog box displays. Select from the list or enter a **Host Name** (the IP address of the MultiVOIP 400). In this example, the Host Name is 192.168.2.8. Then select **Telnet** from the **Port** list and **vt100** from the **Term Type** list.



Once you have entered a valid Host Name (IP address), Port, and Term Type, click **Connect** to connect to the target MultiVOIP. The MultiVOIP Telnet Server window displays.

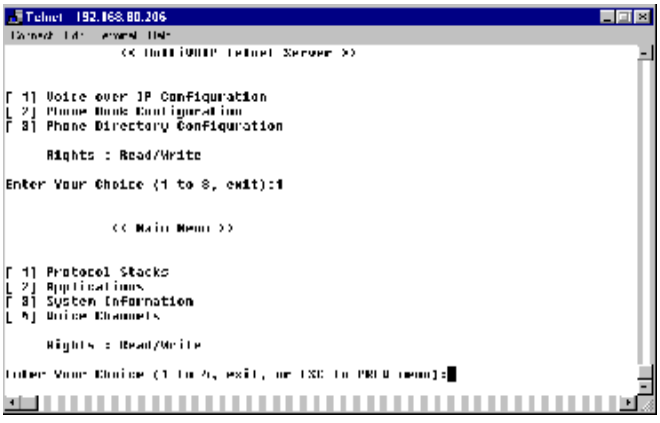
MultiVOIP Telnet Server Menu

The MultiVOIP Telnet Server menu provides three basic options: Voice over IP Configuration, Phone Directory Database, and Phone Directory Configuration.



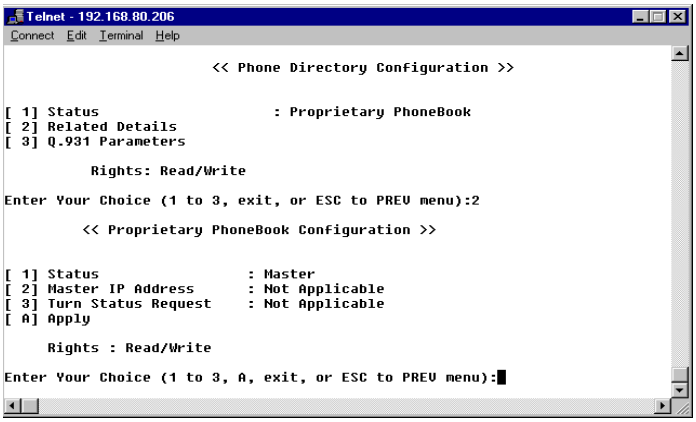
Voice over IP Configuration

Selecting **Option 1** displays the Main menu, which enables further configuration options. These options include Protocol Stacks (option 1), Applications (option 2), System Information (option 3), and Voice Channels (option 4). For further descriptions of these options, refer to Chapter 4 - MultiVOIP 400 Software.



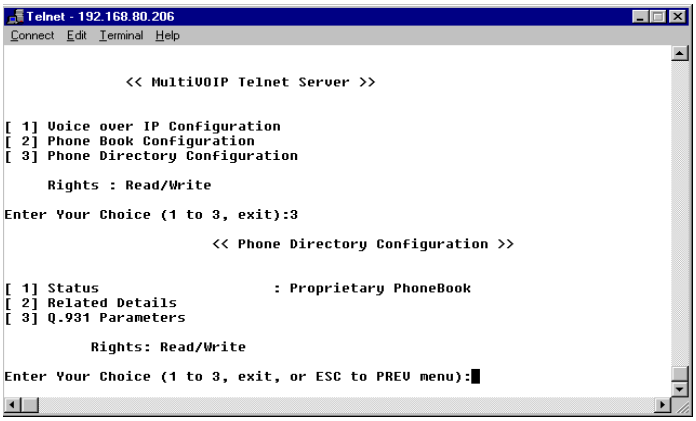
Phone Directory Database

Selecting **Option 2** enables you to add entries to the **Phone Directory Database**. Refer to Chapter 4 - MultiVOIP 400 Software, for more details on the database.



Phone Directory Configuration

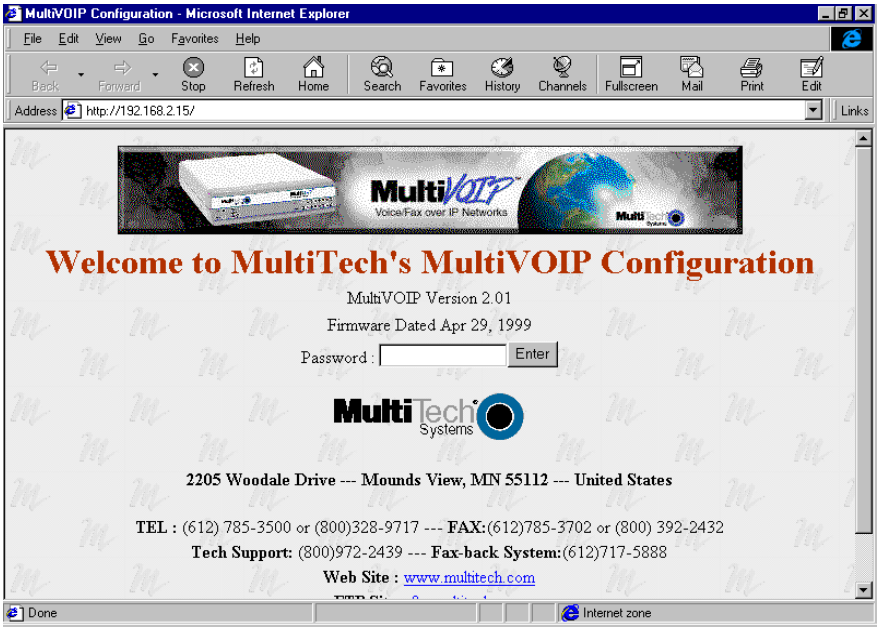
Selecting **Option 3** enables you to configure and manage the **Phone Directory**. The various options are described in detail in Chapter 4 - MultiVOIP 400 Software.



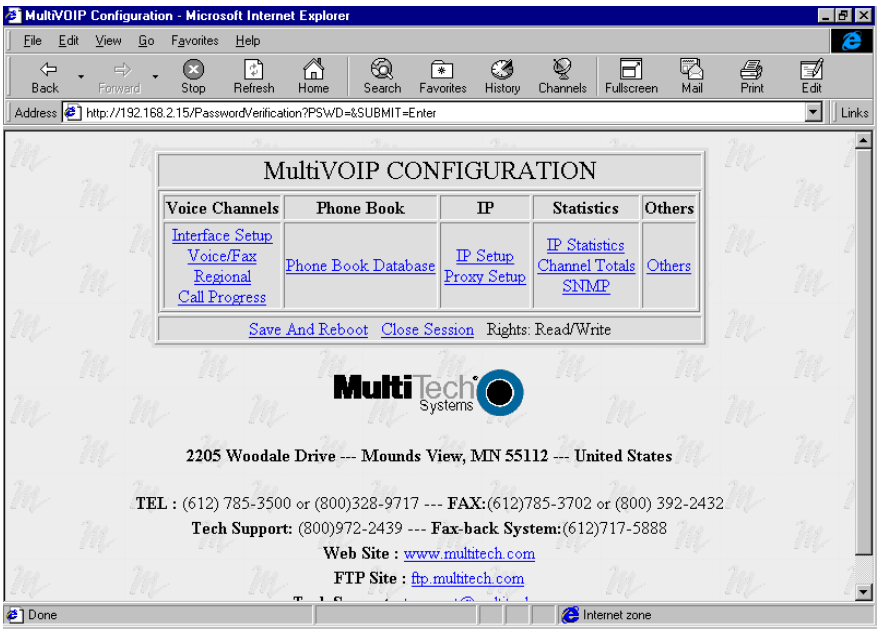
WEB Management

The MultiVOIP 400 can be accessed, via a standard Web browser, from anywhere on the connected Internet. In order to provide this support, the **WEB Server** option has to be selected in the the **Applications Setup** dialog box. Click **Others** in the main menu to open this dialog box. See Chapter 4 - MultiVOIP 400 Software for more information.

Once enabled, you can access the MultiVOIP 400 by entering its IP address in the **Address** box of your Web browser. The following Web page displays.


MultiVOIP Configuration - Microsoft Internet Explorer
File Edit View Go Favorites Help
Back Forward Stop Refresh Home Search Favorites History Channels Fullscreen Mail Print Edit
Address http://192.168.2.15/
Links
MultiVOIP
Voice/Fax over IP Networks
Welcome to MultiTech's MultiVOIP Configuration
MultiVOIP Version 2.01
Firmware Dated Apr 29, 1999
Password :
Enter
MultiTech Systems
2205 Woodale Drive --- Mounds View, MN 55112 --- United States
TEL : (612) 785-3500 or (800)328-9717 --- FAX:(612)785-3702 or (800) 392-2432
Tech Support: (800)972-2439 --- Fax-back System:(612)717-5888
Web Site : www.multitech.com
Done
Internet zone

If a Password was entered in the **Applications Setup** dialog box, enter the password and click **Enter**.



From this Web page you can access all the configuration options. Refer to Chapter 4 - MultiVOIP 400 Software, for a description of the various options.

MultiVOIP™

Voice/Fax over IP Networks

Chapter 6 - Warranty, Service and Tech Support

MultiTech® Systems

Introduction

This chapter begins with statements about your MultiVOIP 400 2-year warranty. The next section, Tech Support, should be read carefully if you have questions or problems with your MultiVOIP 400. It includes the technical support phone numbers, space for recording your product information, and an explanation of how to send in your MultiVOIP 400 should you require service. The final section explains how to receive support from the Internet.

Limited Warranty

Multi-Tech Systems, Inc., (hereafter "MTS") warrants that its products will be free from defects in material or workmanship for a period of two years from date of purchase, or if proof of purchase is not provided, two years from date of shipment.

MTS MAKES NO OTHER WARRANTY, EXPRESS OR IMPLIED, AND ALL IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE HEREBY DISCLAIMED.

This warranty does not apply to any products which have been damaged by lightning storms, water, or power surges or which have been neglected, altered, abused, used for a purpose other than the one for which they were manufactured, repaired by Customer or any party without MTS's written authorization, or used in any manner inconsistent with MTS's instructions.

MTS's entire obligation under this warranty shall be limited (at MTS's option) to repair or replacement of any products which prove to be defective within the warranty period or, at MTS's option, issuance of a refund of the purchase price. Defective products must be returned by Customer to MTS's factory — transportation prepaid.

MTS WILL NOT BE LIABLE FOR CONSEQUENTIAL DAMAGES, AND UNDER NO CIRCUMSTANCES WILL ITS LIABILITY EXCEED THE PRICE FOR DEFECTIVE PRODUCTS.

Addendum for North American Products

In the event that service is required, products may be shipped, freight prepaid, to our Mounds View, Minnesota, factory (**Multi-Tech Systems, Inc., 2205 Woodale Drive, Mounds View, MN 55112, Attn: Repairs, Serial #_____**). A Returned Materials Authorization (RMA) is not required. Return shipping charges (surface) will be paid by MTS. Please include, inside the shipping box, a description of the problem, a return shipping address (must have street address, not P.O. Box), a telephone number, and if the product is out of warranty, a check or purchase order for repair charges.

Extended two-year overnight replacement agreements are available for selected products. Please refer to our Overnight Replacement Agreement for details on rates and coverages. Please direct your questions regarding technical matters, product configuration, and verification that the product is defective to our Technical Support department at **1-800-972-2439**.

Please direct your questions regarding repair expediting, receiving, shipping, and billing to our Repair Accounting department at (800) 328-9717 or (763) 785-3500.

Repairs for damages caused by lightning storms, water, power surges, incorrect installation, physical abuse, or user-caused damages are billed on a time-plus-materials basis.

Addendum for International Products

Distributors should contact Amex, Inc., for information about the repairs for your Multi-Tech product.

Amex, Inc.
2724 Summer Street NE Minneapolis, MN 55413
U.S.A. Tel: (763) 331-3251
Fax: (763) 331-3180

Please direct your questions regarding technical matters, product configuration, verification that the product is defective, etc., to our Technical Support department nearest you. When calling the U.S., please direct your questions regarding repair expediting, receiving, shipping, billing, etc., to our Repair Accounting department at (763) 785-3500 in the U.S.A., or a nearby Multi-Tech office which is listed on the "Multi-Tech Corporate Offices" sheet in this International Distributor Resource Kit.

Repairs for damages caused by lightning storms, water, power surges, incorrect installation, physical abuse, or Out of Warranty Repair Costs

Refer to Multi-Tech System's web site at http://www.multitech.com for information about out of warranty repair costs.ser-caused damages are billed on a time-plus-materials basis.

Service

If your tech support specialist decides that service is required, your MultiVOIP 400 may be sent (freight prepaid) to our factory. Return shipping charges will be paid by Multi-Tech Systems.

Include the following with your MultiVOIP 400:

- a description of the problem
- return billing and return shipping addresses
- contact name and phone number
- check or purchase order number for payment if the MultiVOIP 400 is out of warranty (Check with your technical support specialist for the standard repair charge for your MultiVOIP 400)
- if possible, note the name of the technical support specialist with whom you spoke

If you need to inquire about the status of the returned product, be prepared to provide the **serial number** of the product sent.

Send your MultiVOIP 400 to this address:

MULTI-TECH SYSTEMS, INC.
2205 WOODALE DRIVE
MOUNDS VIEW, MINNESOTA 55112
ATTN: SERVICE OR REPAIRS

You should also check with the supplier of your MultiVOIP 400 on the availability of local service and/or loaner units in your part of the country.

Ordering Accessories

SupplyNet, Inc. supplies replacement transformers, cables and connectors for select Multi-Tech products. You can place an order with SupplyNet via mail, phone, fax or the Internet at:

Mail: SupplyNet, Inc.
614 Corporate Way
Valley Cottage, NY 10989

Phone: 800 826-0279

Fax: 914 267-2420

Email: info@thesupplynet.com

Internet: http://www.thesupplynet.com

Tech Support

Multi-Tech has an excellent staff of technical support personnel available to help you get the most out of your Multi-Tech product. If you have any questions about the operation of this unit, call 1-800-972-2439. Please fill out the MultiVOIP 400 information (below), and have it available when you call. If your MultiVOIP 400 requires service, the tech support specialist will guide you on how to send in your MultiVOIP 400 (refer to the next section).

Recording MultiVOIP 400 Information

Please fill in the following information on your Multi-Tech MultiVOIP. This will help tech support in answering your questions. (The same information is requested on the Warranty Registration Card.)

Model No.: _____

Serial No.: _____

Software Version: _____

The model and serial numbers are on the bottom of your MultiVOIP 400.

Please note status of your MultiVOIP 400 including LED indicators, screen messages, diagnostic test results, problems with a specific application, and so on. Use the space below to note the MultiVOIP 400 status:

_____

_____

_____

_____

_____

_____

About the Internet

Multi-Tech's presence includes a Web site at:

http://www.multitech.com

and an ftp site at:

ftp://ftp.multitech.com

MultiVOIP™

Voice/Fax over IP Networks

Appendixes

MultiTech® Systems

Appendix A - TCP/IP Description

TCP/IP (Transmission Control Protocol/Internet Protocol) is a protocol suite and related applications developed for the U.S. Department of Defense in the 1970s and 1980s specifically to permit different types of computers to communicate and exchange information with one another. TCP/IP is currently mandated as an official U.S. Department of Defense protocol and is also widely used in the UNIX community.

Before you install TCP/IP on your network, you need to establish your Internet addressing strategy. First, choose a domain name for your company. A domain name is the unique Internet name, usually the name of your business, that identifies your company. For example, Multi-Tech's domain name is multitech.com (.com indicates this is a commercial organization; .edu denotes educational organizations, .gov denotes government organizations). Next, determine how many IP addresses you will need. This depends on how many individual network segments you have, and how many systems on each segment need to be connected to the Internet. You will need an IP address for each network interface on each computer and hardware device.

IP addresses are 32 bits long and come in two types: network and host. Network addresses come in five classes: A, B, C, D, and E. Each class of network address is allocated a certain number of host addresses. For example, a class B network can have a maximum of 65,534 hosts, while a class C network can have only 254. The class A and B addresses have been exhausted, and the class D and E addresses are reserved for special use. Consequently, companies now seeking an Internet connection are limited to class C addresses.

Early IP implementations ran on hosts commonly interconnected by Ethernet local area networks (LAN). Every transmission on the LAN contains the local network, or medium access control (MAC), address of the source and destination nodes. The MAC address is 48-bits in length and is non-hierarchical; MAC addresses are never the same as IP addresses.

When a host needs to send a datagram to another host on the same network, the sending application must know both the IP and MAC addresses of the intended receiver. Unfortunately, the IP process may not know the MAC address of the receiver. The Address Resolution Protocol (ARP), described in RFC 826 (located at ftp://ds.internic.net/rfc/rfc826.txt) provides a mechanism for a host to determine a receiver's MAC address from the IP address. In the process, the host sends an ARP packet in a frame containing the MAC broadcast address; and then the ARP request advertises the destination IP address and asks for the associated MAC address. The station on the LAN that recognizes its own IP address will send an ARP response with its own MAC address. An ARP message is carried directly in an IP datagram.

Other address resolution procedures have also been defined, including those which allow a diskless processor to determine its IP address from its MAC address (Reverse ARP, or RARP), provides a mapping between an IP address and a frame relay virtual circuit identifier (Inverse ARP, or InARP), and provides a mapping between an IP address and ATM virtual path/channel identifiers (ATMARP).

The TCP/IP protocol suite comprises two protocols that correspond roughly to the OSI Transport and Session Layers; these protocols are called the Transmission Control Protocol (TCP) and the User Datagram Protocol (UDP). Individual applications are referred to by a port identifier in TCP/UDP messages. The port identifier and IP address together form a "socket". Well-known port numbers on the server side of a connection include 20 (FTP data transfer), 21 (FTP control), 23 (Telnet), 25 (SMTP), 43 (whois), 70 (Gopher), 79 (finger), and 80 (HTTP).

TCP, described in RFC 793 (ftp://ds.internic.net/rfc/rfc793.txt) provides a virtual circuit (connection-oriented) communication service across the network. TCP includes rules for formatting messages, establishing and terminating virtual circuits, sequencing, flow control, and error correction. Most of the applications in the TCP/IP suite operate over the "reliable" transport service provided by TCP.

UDP, described in RFC 768 (ftp://ds.internic.net/rfc/rfc768.txt) provides an end-to-end datagram (connectionless) service. Some applications, such as those that involve a simple query and response, are better suited to the datagram service of UDP because there is no time lost to virtual

circuit establishment and termination. UDP's primary function is to add a port number to the IP address to provide a socket for the application.

The Application Layer protocols are examples of common TCP/IP applications and utilities, which include:

- Telnet (Telecommunication Network): a virtual terminal protocol enabling a user logged on to one TCP/IP host to access other hosts on the network, described in RFC 854 (ftp://ds.internic.net/rfc/rfc854.txt).
- FTP: the File Transfer Protocol enables a user to transfer files between local and remote host computers per IETF RFC 959 (ftp://ds.internic.net/rfc/rfc959.txt).
- Archie: a utility that enables a user to search all registered anonymous FTP sites for files on a specified topic.
- Gopher: a tool that enables users to search through data repositories using a menu-driven, hierarchical interface, with links to other sites, per RFC 1436 (ftp://ds.internic.net/rfc/rfc1436.txt).
- SMTP: the Simple Mail Transfer Protocol is the standard protocol for the exchange of electronic mail over the Internet, per IETF RFC 821 (ftp://ds.internic.net/rfc/rfc821.txt).
- HTTP: the Hypertext Transfer Protocol is the basis for exchange of information over the World Wide Web (WWW). Various versions of HTTP are in use over the Internet, with HTTP version 1.0 (per RFC 1945) (ftp://ds.internic.net/rfc/rfc1945.txt) being the most current.
- HTML: WWW pages are written in the Hypertext Markup Language (HTML), an ASCII-based, platform-independent formatting language, per IETF RFC 1866 (ftp://ds.internic.net/rfc/rfc1866.txt).
- Finger: used to determine the status of other hosts and/or users, per IETF RFC 1288 (ftp://ds.internic.net/rfc/rfc1288.txt).
- POP: the Post Office Protocol defines a simple interface between a user's mail reader software and an electronic mail server; the current version is POP3, described in IETF RFC 1460 (ftp://ds.internic.net/rfc/rfc1460.txt).
- DNS: the Domain Name System defines the structure of Internet names and their association with IP addresses, as well as the association of mail, name, and other servers with domains.
- SNMP: the Simple Network Management Protocol defines procedures and management information databases for managing TCP/IP-based network devices. SNMP, defined by RFC 1157 (ftp://ds.internic.net/rfc/rfc1157.txt) is widely deployed in local and wide area network. SNMP Version 2 (SNMPv2), per RFC 1441< ftp://ds.internic.net/rfc/rfc1441.txt) adds security mechanisms that are missing in SNMP, but is also more complex.
- Ping: a utility that enables a user at one system to determine the status of other hosts and the latency in getting a message to that host. Ping uses ICMP Echo messages.
- Whois/NICNAME: Utilities that search databases for information about Internet domain and domain contact information, per RFC 954 (ftp://ds.internic.net/rfc/rfc954.txt).
- Traceroute: a tool that displays the route that packets will take when traveling to a remote host.

Internet Protocol (IP)

IP is the Internet standard protocol that tracks Internetwork node addresses, routes outgoing messages and recognizes incoming messages, allowing a message to cross multiple networks on the way to its final destination. The IPv6 Control Protocol (IPV6CP) is responsible for configuring, enabling, and disabling the IPv6 protocol modules on both ends of the point-to-point link. IPV6CP uses the same packet exchange mechanism as the Link Control Protocol (LCP). IPV6CP packets are not exchanged until PPP has reached the Network-Layer Protocol phase. IPV6CP packets received before this phase is reached are silently discarded. (See also TCP/IP.)

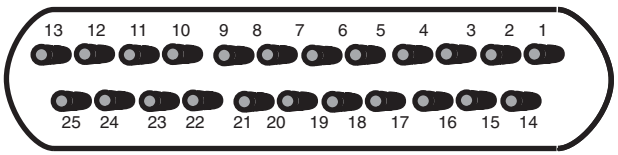
Before you install TCP/IP on your network, you need to establish your Internet addressing strategy. You first choose a domain name for your company. A domain name is the unique Internet name, usually the name of your business, that identifies your company. For example, Multi-Tech's domain name is multitech.com (where .com indicates this is a commercial organization; .edu denotes educational organizations, .gov denotes government organizations). Next, you determine how many IP addresses you will need. This depends on how many individual network segments you have, and how many systems on each segment need to be connected to the Internet. You need an IP address for each network interface on each computer and hardware device.

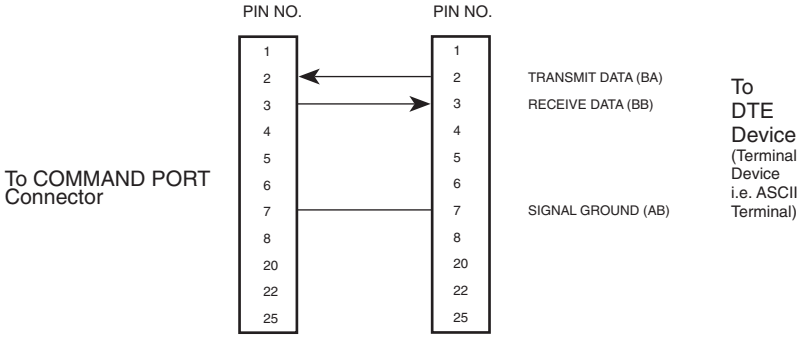
IP addresses are 32 bits long and come in two types: network and host. Network addresses come in five classes: A, B, C, D, and E. Each class of network address is allocated a certain number of host addresses. For example, a class B network can have a maximum of 65,534 hosts, while a class C network can have only 254. The class A and B addresses have been exhausted, and the class D and E addresses are reserved for special use. Consequently, companies now seeking an Internet connection are limited to class C addresses. The current demand for Internet connections will exhaust the current stock of 32-bit IP addresses. In response, Internet architects have proposed the next generation of IP addresses, Ipng (IP Next Generation). It features 16-byte addressing, surpassing the capacities of 32-bit IP.

An IP address can serve only a single physical network. Therefore, if your organization has multiple physical networks, you must make them appear as one to external users. This is done via "subnetting", a complex procedure best left to ISPs and others experienced in IP addressing. Since IP addresses and domain names have no inherent connection, they are mapped together in databases stored on Domain Name Servers (DNS). If you decide to let an Internet Service Provider (ISP) administer your DNS server, the ISP can assist you with the domain name and IP address assignment necessary to configure your company's site-specific system information. Domain names and IP addresses are granted by the InterNIC. To check the availability of a specific name or to obtain more information, call the InterNIC at (703)742-4777.

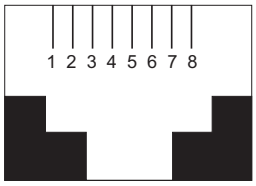
Appendix B - Cabling Diagrams

Command Port Cable



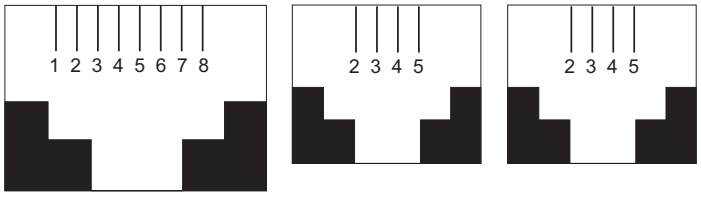

PIN NO.
PIN NO.
1 2 3 4 5 6 7 8 20 22 25
1 2 3 4 5 6 7 8 20 22 25
TRANSMIT DATA (BA)
RECEIVE DATA (BB)
SIGNAL GROUND (AB)
To COMMAND PORT Connector
To DTE Device (Terminal Device i.e. ASCII Terminal)

LAN Cable



Pin	Circuit Signal Name
1	TD+ Data Transmit Positive
2	TD- Data Transmit Negative
3	RD+ Data Receive Positive
6	RD- Data Receive Negative

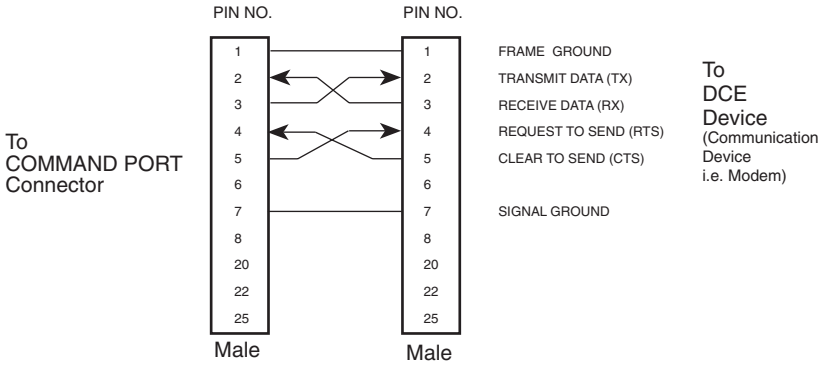
Voice/Fax Channel Connectors



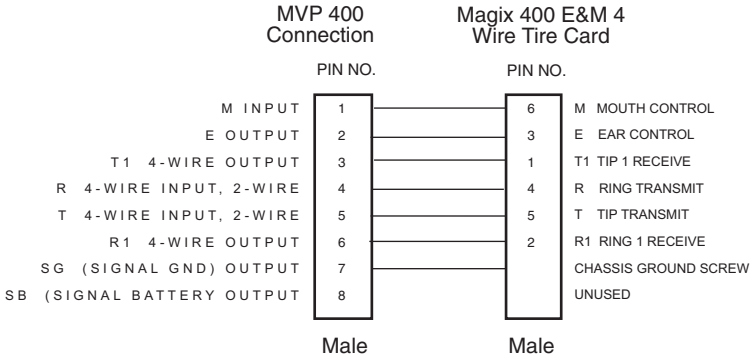
Pin Connections

E&M	Desc.	Function	FXO	Description	FXS	Description
1	M	Input				
2	E	Output	2	N/C	2	N/C
3	T1	4-Wire Output	3	Ring	3	Tip
4	R	4-Wire Input, 2-Wire	4	Tip	4	Ring
5	T	4-Wire Input, 2-Wire	5	N/C	5	N/C
6	R1	4-Wire Output				
7	SG	(Signal Ground) Output				
8	SB	(Signal Battery) Output				

Remote Configuration Cable



Magix 400 E&M Tie Card



Appendix C - Regulatory Information

Class A Statement

FCC Part 15

NOTE: This equipment has been tested and found to comply with the limits for a **Class A** digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference, in which case the user will be required to correct the interference at his own expense.

This device complies with Part 15 of the FCC rules.
Operation is subject to the following two conditions:

(1) This device may not cause harmful interference.

(2) This device must accept any interference that may cause undesired operation.

Warning: Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

Industry Canada

This **Class A** digital apparatus meets all requirements of the Canadian Interference-Causing Equipment Regulations.

Cet appareil numerique de la classe A respecte toutes les exigences du Reglement sur le materiel brouilleur du Canada.

Fax Branding Statement

The Telephone Consumer Protection Act of 1991 makes it unlawful for any person to use a computer or other electronic device, including fax machines, to send any message unless such message clearly contains the following information:

- Date and time the message is sent
- Identification of the business or other entity, or other individual sending the message
- Phone number of the sending machine or such business, other entity, or individual

This information is to appear in a margin at the top or bottom of each transmitted page or on the first page of the transmission. (Adding this information in the margin is referred to as *fax branding*.)

Since any number of Fax software packages can be used with this product, the user must refer to the Fax software manual for setup details. Typically, the Fax branding information must be entered via the configuration menu of the software.

FCC Part 68 Telecom

1. This equipment complies with Part 68 of the Federal Communications Commission (FCC) rules. On the outside surface of this equipment is a label that contains, among other information, the FCC registration number and ringer equivalence number (REN). If requested, this information must be provided to the telephone company.
2. As indicated below, the suitable jack (Universal Service Order Code connecting arrangement) for this equipment is shown. If applicable, the facility interface codes (FIC) and service order codes (SOC) are shown. An FCC-compliant telephone cord and modular plug is provided with this equipment. This equipment is designed to be connected to the telephone network or premises wiring using a compatible modular jack which is Part 68 compliant. See installation instructions for details.
3. The ringer equivalence number (REN) is used to determine the number of devices which may be connected to the telephone line. Excessive REN's on the telephone line may result in the devices not ringing in response to an incoming call. In most, but not all areas, the sum of the REN's should not exceed five (5.0). To be certain of the number of devices that may be connected to the line, as determined by the total REN's, contact the telephone company to determine the maximum REN for the calling area.
4. If this equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.
5. The telephone company may make changes in its facilities, equipment, operations, or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make necessary modifications in order to maintain uninterrupted service.
6. If trouble is experienced with this equipment (the model of which is indicated below) please contact Multi-Tech Systems, Inc., at the address shown below for details of how to have repairs made. If the equipment is causing harm to the telephone network, the telephone company may request that you remove the equipment from the network until the problem is resolved.
7. No repairs are to be made by you. Repairs are to be made only by Multi-Tech Systems or its licensees. Unauthorized repairs void registration and warranty.
8. This equipment cannot be used on public coin service provided by the telephone company. Connection to Party Line Service is subject to state tariffs. (Contact the state public utility commission, public service commission or corporation commission for information.)
9. If so required, this equipment is hearing-aid compatible.

Manufacturer:	Multi-Tech Systems, Inc.
Trade name:	MultiVOIP
Model Numbers:	MVP400
FCC Registration Number:	AU7USA-26050-DV-N
Modular Jack (USOC):	RJ-11C or RJ-11W
Service Center in U.S.A.:	Multi-Tech Systems Inc. 2205 Woodale Drive Mounds View, MN 55112 (763) 785-3500 Fax (763) 785-9874

Canadian Limitations Notice

Ringer Equivalence Number

Notice: The ringer equivalence number (REN) assigned to each terminal device provides an indication of the maximum number of terminals allowed to be connected to a phone interface. The termination on an interface may consist of any combination of devices subject only to the requirement that the sum of the ringer equivalence numbers of all the devices does not exceed 5.

Notice: The Industry Canada label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational, and safety requirements. The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations. Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, phone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

Caution: Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

CE

EMC, Safety and Terminal Directive Compliance

The CE mark is affixed to this product to confirm compliance with the following European Community Directives:

Council Directive 89/336/EEC of 3 May 1989 on the approximation of the laws of Member States relating to electromagnetic compatibility.

and

Council Directive 73/23/EEC of 19 February 1973 on the harmonization of the laws of Member States relating to electrical equipment designed for use within certain voltage limits:

and

Council Directive 98/13/EC of 12 March 1998 on the approximation of the laws of Member States concerning telecommunications terminal and Satellite earth station equipment.

MultiVOIP™

Voice/Fax over IP Networks

Glossary

MultiTech® Systems

A

Access: The T1 line element made up of two pairs of wire that the phone company brings to the customer premises. The Access portion ends with a connection at the local telco (LEC or RBOC).

Accunet Spectrum of Digital Services (ASDS): The AT&T 56 Kbps leased (private) line service. Similar to services of MCI and Sprint. ASDS is available in *n*x56/64 Kbps, where *n*=1, 2, 4, 6, 8, 12.

ACK (ACKnowledgement code) (pronounced "ack"): A communications code sent from a receiving modem to a transmitting modem to indicate that it is ready to accept data. It is also used to acknowledge the error-free receipt of transmitted data. Contrast with NAK.

Adaptive Differential Pulse Code (ADCPM): In multimedia applications, a technique in which pulse code modulation samples are compressed before they are stored on a disk. ADCPM, an extension of the PCM format, is a standard encoding format for storing audio information in a digital format. It reduced storage requirements by storing differences between successive digital samples rather than full values.

Address: A numbered location inside a computer. This is how the computer accesses resources such as a video card, serial ports, and memory.

AMI line coding: One of two common methods of T1 line coding (with B8ZS). AMI line coding places restrictions on user data (B8ZS does not).

Analog signal: A waveform which has amplitude, frequency and phase, and which takes on a range of values between its maximum and minimum points.

Analog Transmission: One of two types of telecommunications which uses an analog signal as a carrier of voice, data, video, and so on. An analog signal becomes a carrier when it is modulated by altering its phase, amplitude, and frequency to correspond with the source signal. Compare with digital transmission.

Application Program Interface (API): A software module created to enable dissimilar, or incompatible applications programs to transfer information over a communications link. APIs may be simple or complex; they are commonly required to link PC applications with mainframe programs.

ASCII (American Standard Code for Information Interchange) (pronounced "askey"): A binary code for data that is used in communications and in many computers and terminals. The code is used to represent numbers, letters, punctuation and control characters. The basic ASCII code is a 7-bit character set which defines 128 possible characters. The extended ASCII file provides 255 characters.

Asynchronous Transfer Mode (ATM): A very high-speed method of transmission that uses fixed-size cells of 53 bytes to transfer information over fiber; also known as cell relay.

AT Commands: A standard set of commands used to configure various modem parameters, establish connections and disconnect. The "AT" is used to get the "attention" of the modem before the actual command is issued.

Availability: The measure of the time during which a circuit is ready for use; the complement of circuit "outage" (100% minus % outage = % available).

B

B7ZS (Bipolar 7 Zero Suppression) line coding: One method of T1 line coding (see also "B8ZS" and "AMI"). B7ZS line coding does not place restrictions on user data (AMI does).

B8ZS (Bipolar 8 Zero Suppression) line coding: One of two common methods of T1 line coding (with AMI). B8ZS line coding does not place restrictions on user data (AMI does). A coding method used to produce 64 Kbps "clear" transmission. See also "B7ZS" and "AMI" line coding.

Backbone: 1) A set of nodes and their interconnecting links providing the primary data path across a network. 2) In a local area network multiple-bridge ring configuration, a high-speed link to which the rings are connected by means of bridges. A backbone may be configured as a bus or as a ring. 3) In a wide area network, a high-speed link to which nodes or data switching exchanges (DSEs) are connected. 4) A common distribution core that provides all electrical power, gases, chemicals, and other services to the sectors of an automated wager processing system.

Background: An activity that takes place in the PC while you are running another application. In other words, the active user interface does not correspond to the 'background' task.

Bandwidth: The transmission capacity of a computer channel, communications line or bus. It is expressed in cycles per second (hertz), the bandwidth being the difference between the lowest and highest frequencies transmitted. The range of usable frequencies that a transmission medium will pass without unacceptable attenuation or distortion. Bandwidth is a factor in determining the amount of information and the speed at which a medium can transmit data or other information.

Backward Explicit Congestion Notification (BECN): A bit that tells you that a certain frame on a particular logical connection has encountered heavy traffic. The bit provides notification that congestion-avoidance procedures should be initiated in the opposite direction of the received frame. See also FECN (Forward Explicit Congestion Notification).

Basic Rate Interface (BRI): An ISDN access interface type comprised of two B-channels each at 64 Kbps and one D-channel at 64 Kbps (2B+D).

Bell Operating Companies (BOC): The family of corporations created during the divestiture of AT&T. BOCs are independent companies which service a specific region of the US. Also called Regional Bell Operating Companies (RBOCs).

Bell Pub 41450: The Bell publication defining requirements for data format conversion, line conditioning, and termination for direct DDS connection.

Bell Pub 62310: The Bell publication defining requirements for data format conversion, line conditioning, and termination for direct DDS connection.

Binary Synchronous Communication (BSC): A form of telecommunication line control that uses a standard set of transmission control characters and control character sequences, for binary synchronous transmission of binary-coded data between stations.

Bit (Binary digIT): A bit is the basis of the binary number system. It can take the value of 1 or 0. Bits are generally recognized as the electrical charge generated or stored by a computer that represent some portion of usable information.

Bit Error Rate Test (BERT): A device or routine that measures the quality of data transmission. A known bit pattern is transmitted, and the errors received are counted and a BER (bit error rate) is calculated. The BER is the ratio of received bits in error relative to the total number of bits received, expressed in a power of 10.

Bit robbing: The use of the least significant bit per channel in every sixth frame for signaling. The line signal bits "robbed" from the speech pat conveys sufficient pre-ISDN telephony signaling information with the remaining line signal bits providing sufficient line signaling bits for recreating the original sound. See "robbed bit signaling".

Blue Alarm: An error indication signal consisting of all 1s indicating disconnection or attached device failure. Contrast "Red Alarm" and "Yellow Alarm".

Bps (bits per second): A unit to measure the speed at which data bits can be transmitted or received. Bps differs from baud when more than one bit is represented by a single cycle of the carrier.

Bridges: 1) A functional unit that interconnects two local area networks that use the same logical link protocol but may use different medium access control protocols. 2) A functional unit that interconnects multiple LANs (locally or remotely) that use the same logical link control protocol but that can use different medium access control protocols. A bridge forwards a frame to another bridge based on the medium access control (MAC) address. 3) In the connection of local loops, channels, or rings, the equipment and techniques used to match circuits and to facilitate accurate data transmission.

Buffer: A temporary storage register or Random Access Memory (RAM) used in all aspects of data communications which prevents data from being lost due to differences in transmission speed. Keyboards, serial ports, muxes and printers are a few examples of the devices that contain buffers.

Bus: A common channel between hardware devices either internally between components in a computer, or externally between stations in a communications network.

Byte: The unit of information a computer can handle at one time. The most common understanding is that a byte consists of 8 binary digits (bits), because that's what computers can handle. A byte holds the equivalent of a single character (such as the letter A).

C

Call Setup Time: The time to establish a circuit-switched call between two points. Includes dialing, wait time, and CO/long distance service movement time.

Carrier Group Alarm (CGA): A T1 service alarm generated by a channel bank when an OOF condition occurs for a predefined length of time (usually 300mS to 2.5 seconds). The CGA causes the calls using a trunk to be dropped and for trunk conditioning to be applied.

Carrier signal: An analog signal with known frequency, amplitude and phase characteristics used as a transport facility for useful information. By knowing the original characteristics, a receiver can interpret any changes as modulations, and thereby recover the information.

CCITT (Consultative Committee for International Telephone and Telegraph): An advisory committee created and controlled by the United Nations and headquartered in Geneva whose purpose is to develop and to publish recommendations for worldwide standardization of telecommunications devices. CCITT has developed modem standards that are adapted primarily by PTT (post, telephone and telegraph) organizations that operate telephone networks of countries outside of the U.S. See also ITU.

Central Office (CO): The lowest, or most basic level of switching in the PSTN (public switched telephone network). A business PABX or any residential phone connects to the PSTN at a central office.

Centrex: A multi-line service offered by operating telcos which provides, from the telco CO, functions and features comparable to those of a PBX for large business users. See also "Private Branch Exchange" and "Exchange".

Channel: A data communications path between two computer devices. Can refer to a physical medium (for example, UTP or coax), or to a specific carrier frequency.

Channel bank: A device that acts as a converter, taking the digital signal from the T1 line into a phone system and converting it to the analog signals used by the phone system. A channel bank acts as a multiplexer, placing many slow-speed voice or data transactions on a single high-speed link.

Circuit-switched Network: A technology used by the PSTN that allocates a pair of conductors for the exclusive use of one communication path. Circuit switching enables multiple conversations on one talk path only if the end-users multiplex the signals prior to transmission.

Circuit switching: The temporary connection of two or more communications channels using a fixed, non-shareable path through the network. Users have full use of the circuit until the connection is terminated.

Clear Channel: A transmission path where the full bandwidth is used (no bandwidth needed for signaling, carrier framing or control bits). A 64 Kbps digital circuit usually has 8 Kbps used for signaling. ISDN has two 64 Kbps circuits, and a 16 Kbps packet service of which part is used for signaling on the 64K channels.

Client-Server: In TCP/IP, the model of interaction in distributed data processing in which a program at one site sends a request to a program at another site and awaits a response. The requesting program is called a client; the answering program is called a server.

Cluster Controller: A device that can control the input/output operations of more than one device connected to it. A cluster controller may be controlled by a program stored and executed in the unit, or it may be entirely controlled by hardware.

CODEC (COmpression/DEcompression): The term is used to describe the conversion of voice signals from their analog form to digital signals acceptable to modern digital PBXs and digital transmission systems. It then converts those digital signals back to analog so that you can hear and understand what the other person is saying. In some phone systems, the CODEC is in the PBX and shared by many analog phone extensions. In other phone systems, the CODEC is actually in the phone. Thus the phone itself sends out a digital signal and can, as a result, be more easily designed to accept a digital RS-232-C signal.

Committed Burst Size: The maximum number of bits that the frame relay network agrees to transfer during any measurement interval

Committed Information Rate (CIR): An agreement a customer makes to use a certain minimum data transmission rate (in bps). The CIR is part of the frame relay service monthly billing, along with actual usage, that users pay to their frame relay service provider.

Compression: 1) The process of eliminating gaps, empty boxes, redundancies, and unnecessary data to shorten the length of records or blocks. 2) In SNA, the replacement of a string of up to 64-repeated characters by an encoded control byte to reduce the length of the data stream to the LU-LU session partner. The encoded control byte is followed by the character that was repeated, unless that character is the prime compression character. 3) In Data Facility Hierarchical Storage Manager, the process of moving data instead of allocated space during migration and recall in order to release unused space. 4) Contrast with decompression.

COMx Port: A serial communications port on a PC.

Congestion: A network condition where there is too much data traffic. The ITU I.233 standard defines congestion management in terms of speed and burstiness.

Congestion notification: The function in frame relay that ensures that user data transmitted at a rate higher than the CIR is slowed to the rate of the available network bandwidth.

Consecutive Severely Errored Seconds (CSES): An error condition that occurs when from 3 to 9 SES (Severely Errored Seconds) are logged consecutively.

Customer Premise Equipment (CPE): The generic term for data comm and/or terminal equipment that resides at the user site and is owned by the user with the following exclusions: over voltage protection equipment, inside wiring, coin operated or pay telephones, "company-official" equipment, mobile phone equipment, "911" equipment, equipment necessary for the provision of communications for national defense, or multiplexing equipment used to deliver multiple channels to the customer.

D

D4: the T1 4th generation channel bank.

D4 channelization: Refers to the compliance with AT&T TR 62411 for DS1 frame layout.

D4 framing: The T1 format for framing in AT&T D-Series channel banks, in which there are 12 separate 193-bit frames in a super-frame. A D4 framing bit is used to identify the channel and the signaling frame. Signalling for voice channels is carried in-band for every channel, along with the encoded voice. See "robbed-bit signaling".

Data Communications Equipment (DCE): Any device which serves as the portal of entry from the user equipment to a telecommunications facility. A modem is a DCE for the phone network (PSTN) that is commonly on site at the user's premises. Packet Switched Networks have another level of DCE which is most often located at a central office.

Data Link Connection Identifier (DLCI): One of the six components of a frame relay frame. Its purpose is to distinguish separate virtual circuits across each access connection. Data coming into a frame relay node is thus allowed to be sent across the interface to the specified "address". The DLCI is confirmed and relayed to its destination, or if the specification is in error, the frame is discarded.

Dataphone Digital Service (DDS): A private line digital service that offers 2400, 4800, 9600 and 56 Kbps data rates on an inter-LATA basis by AT&T and on an intra-LATA basis by the BOCs.

Data Service Unit (DSU): A device that provides a digital data service interface directly to the data terminal equipment. The DSU provides loop equalization, remote and local testing capabilities, and a standard EIA/CCITT interface.

Dedicated Line: A communication line that is not switched. The term leased line is more common.

Default: This is a preset value or option in software packages, or in hardware configuration, that is used unless you specify otherwise.

Device driver: Software that controls how a computer communicates with a device, such as a printer or mouse.

Digital Cross-connect System (DCS): The CO device which splits and redistributes the T1 bandwidth. The DCS takes time slots from various T1 lines and alters them to provide the needed connectivity. DCS connections are made with software at an administrator's workstation.

Digital Data: Information represented by discrete values or conditions (contrast "Analog Data").

Digital Loopback: A technique used for testing the circuitry of a communications device. Can be initiated locally, or remotely (via a telecommunications device). The tested device decodes and encodes a received test message, then echoes the message back. The results are compared with the original message to determine if corruption occurred en route.

Digital PBX: A Private Branch Exchange that operates internally on digital signals. See also "Exchange".

Digital Service, level 0 (DS0): The world-wide standard speed (64 Kbps) for digital voice conversation using PCM (pulse coded modulation).

Digital Service, level 1 (DS1): The 1.544 Mbps voice standard (derived from an older Bell System standard) for digitized voice transmission in North America. The 1.544 Mbps consists of 24 digitally-encoded 64 Kbps voice channels (north America) and 2.048 Mbps (30 channels) elsewhere.

Digital Signal: A discrete or discontinuous signal (for example, a sequence of voltage pulses). Digital devices, such as terminals and computers, transmit data as a series of electrical pulses which have discrete jumps rather than gradual changes.

Digital Signaling Rates (DSn): A hierarchical system for transmission rates, where "DS0" is 64 Kbps (equivalent to ISDN B channel), and DS1 is 1.5 Mbps (equivalent to ISDN PRI).

Digital Transmission: A method of electronic information transmission common between computers and other digital devices. Analog signals are waveforms: a combination of many possible voltages. A computer's digital signal may be only "high" or "low" at any given time. Therefore, digital signals may be "cleaned up" (noise and distortion removed) and amplified during transmission.

Digitize: To convert an analog signal to a digital signal.

DIP switch (pronounced "dip switch"): A set of tiny toggle switches, built into a DIP (dual in-line package), used for setting configurable parameters on a PCB (printed circuit board).

Driver: A software module that interfaces between the Operating System and a specific hardware device such as color monitors, printers, and hard disks. Also known as a device driver.

Drop and Insert: The process where a portion of information carried in a transmission system is demodulated ("Dropped") at an intermediate point and different information is included ("Inserted") for subsequent transmission.

DTE (Data Terminating Equipment): A term used to include any device in a network which generates, stores, or displays user information. DTE is a telecommunications term which usually refers to PCs, terminals, printers, and so on.

DTMF (Dual-Tone MultiFrequency): A generic push-button concept made popular by AT&T Touch-Tone.

E

E&M: A telephony trunking system used for either switch-to-switch, or switch-to-network, or computer/telephone system-to-switch connection.

EIA: The Electronics Industries Association is a trade organization in Washington, DC that sets standard for use of its member companies. See RS-232, RS-422, RS530.

Encapsulation: A technique used by network-layer protocols in which a layer adds header information to the protocol data unit from the preceding layer. Also used in "enveloping" one protocol inside another for transmission. For example, IP inside IPX.

Endpoint: The clients in an H.323 network. They are typically video conferencing, audio conferencing, or other multimedia systems implemented by end users to communicate in real time. The H.323 standard requires that every endpoint support certain functions and codecs (Coder/Decoder) that have previously been defined by the ITU.

Errored Seconds (ES): Any second of operation that all 1.544M bits are not received exactly as transmitted. Contrast "Error Free Seconds".

Error Free Seconds (EFS): Any second of operation that all 1.544M bits are received exactly as transmitted. Contrast "Errored Seconds".

ESF Error Event: A T1 error condition that is logged when a CRC-6 error or an OOF error occurs.

Ethernet: A 10-megabit baseband local area network that enables multiple stations to access the transmission medium at will without prior coordination, avoids contention by using carrier sense and deference, and resolves contention by using collision detection and transmission. Ethernet uses carrier sense multiple access with collision detection (CSMA/CD).

Excess Zeros: A T1 error condition that is logged when more than 15 consecutive 0s or less than one 1 bit in 16 bits occurs.

Exchange: A unit (public or private) that can consist of one or more central offices established to serve a specified area. An exchange typically has a single rate of charges (tariffs) that has previously been approved by a regulatory group.

Exchange Area: A geographical area with a single uniform set of charges (tariffs), approved by a regulatory group, for phone services. Calls between any two points within an exchange area are local calls. See also "Digital PBX", "PBX".

Exchange Termination (ET): The carrier's local exchange switch. Contrast with "Loop Termination - LT".

Explicit Congestion Management: The method used in frame relay to notify the terminal equipment that the network is very busy. The use of FECN and BECN is called explicit congestion management. Some end-to-end protocols use FECN or BECN, but usually not both options together. With this method, a congestion condition is identified and fixed before it becomes critical. Contrast with "implicit congestion".

Extended Super Frame (ESF): One of two popular formats for framing bits on a T1 line. ESF framing has a 24-frame super-frame, where robbed bit signaling is inserted in the LSB (bit 8 of the DS-0 byte) of frames 6, 12, 18 and 24. ESF has more T1 error measurement capabilities than D4 framing. ESF and B8ZS are typically both offered to provide clear channel service.

F

Failed Seconds: A test parameter where the circuit is unavailable for one full second.

Failed Signal: A T1 test parameter logged when there are more than 9 SES (Severely Errored Seconds).

Fax (facsimile): Refers to the bit-mapped rendition of a graphics-oriented document (fax) or to the electronic transmission of the image over phone lines (faxing). Fax transmission differs from data transmission in that the former is a bit-mapped approximation of a graphical document and, therefore, cannot be accurately interpreted according to any character code.

Firmware: A category of memory chips that hold their content without electrical power, they include ROM, PROM, EPROM and EEPROM technologies. Firmware becomes "hard software" when holding program code.

Foreground: The application program currently running on and in control of the PC screen and keyboard. The area of the screen that occupies the active window. Compare with "background".

Fractional T1 (FT1): A digital data transmission rate between 56 Kbps (DS0 rate) and 1.544 Mbps (the full T1 rate - in North America). FT1 is typically provided on 4-wire (two copper pairs) UTP. Often used for video conferencing, imaging and LAN interconnection due to its low cost and relatively high speed. FT1 rates are offered in 64 Kbps multiples, usually up to 768 Kbps.

Frequency: A characteristic of an electrical or electronic signal which describes the periodic recurrence of cycles. Frequency is inversely proportional to the wavelength or pulse width of the signal (long wavelength signals have low frequencies and short wavelength signals yield high frequencies).

Foreign Exchange (FX): A CO trunk with access to a distant CO, enabling ease of access and flat-rate calls anywhere in the foreign exchange area.

Foreign Exchange Office (FXO): Provides local phone service from a CO outside of ("foreign" to) the subscriber's exchange area. In simple form, a user can pick up the phone in one city and receive a tone in the foreign city.

Connecting a POTS phone to a computer telephony system via a T1 link requires a channel bank configured for the FX connection. To generate a call from the POTS set to the computer telephony system, an FXO connection must be configured.

Foreign Exchange Station (FXS): See FX, FXO. To generate a call from the computer telephony system to the POTS set, a FXS connection must be configured.

Forward Explicit Congestion Notification (FECN): A bit that tells you that a certain frame on a particular logical connection has encountered heavy traffic. The bit provides notification that congestion-avoidance procedures should be initiated in the same direction of the received frame. See also BECN (Backward Explicit Congestion Notification).

Frame: A group of data bits in a specific format to help network equipment recognize what the bits mean and how to process them. The bits are sent serially, with a flag at each end signifying the start and end of the frame.

Frame Relay: A form of packet switching that uses small packets and that requires less error checking than other forms of packet switching. Frame relay is effective for sending "bursty" data at high speeds (56/64K, 256K, and 1024 Kbps) over wide area networks. Frame Relay specifications are defined by ANSI documents ANSI T1.602, T1.606, T1S1/90-175, T1S1/90-213, and T1S1/90-214. In using frame relay, blocks of information (frames) are passed across a digital network interface using a "connection number" that is applied to each frame to distinguish between individual frames.

Frame Relay Forum: A nonprofit organization of 300+ vendors and service providers, based in Foster City, CA, that are developing and deploying frame relay equipment.

Frame Relay Implementors Forum: A group of companies supporting a common specification for frame relay connection to link customer premises equipment to telco network equipment. Their specification supports ANSI frame relay specs and defines extensions such as local management.

Frame Relay Access Device (FRAD): A piece of equipment that acts as a concentrator or frame assembler/dissassembler that can support multiple protocols and provide basic "routing" functions.

G

Gatekeeper: An H.323 entity that provides address translation, control access, and sometimes bandwidth management to the LAN for H.323 endpoints.

Gateway: 1) A functional unit that interconnects two computer networks with different network architectures. A gateway connects networks or systems of different architectures. A bridge interconnects networks or systems with the same or similar architectures. 2) A network that connects hosts. 3) An H.323 entity that provides real-time, two-way communications between H.323 terminals on the LAN and other ITU terminals on a WAN, or to another H.323 Gateway.

Graphical User Interface (GUI): A type of computer interface consisting of a visual metaphor of a real-world scene, often of a desktop. Within that scene are icons, representing actual objects, that the user can access and manipulate with a pointing device.

H

H.323: An umbrella recommendation from the International Telecommunications Union (ITU) that sets standards for multimedia communications over Local Area Networks (LANs) that do not provide a guaranteed Quality of Service (QoS).

H.323 Endpoint: The clients in an H.323 network. They are typically video conferencing, audio conferencing, or other multimedia systems implemented by end users to communicate in real time. The H.323 standard requires that every endpoint support the G.711 ITU standard for speech compression, H.245 protocol for controlling media between H.323 endpoints, Q.931 signal protocol for establishing and terminating calls, RAS (Registration/Admissions/Status) channel data stream used to communicate with a gatekeeper (optional), and RTP/RTCP (Real-Time Protocol/Real-Time Control Protocol) for carrying packetized real-time media on IP networks.

H.323 Entity: Any H.323 component, including terminals, Gateways, Gatekeepers, Multipoint Controllers (MCs), Multipoint Processors (MPs), and Multipoint Control Units (MCUs).

Handshaking: A process that two modems go through at the time of call setup to establish synchronization over the data communications link. It is a synchronization and negotiation process accomplished by the exchange of predefined, mutually recognized control codes.

High-level Data Link Control (HDLC): An ISO standard, bit-oriented data communications protocol that provides nearly error-free data transfers.

I

Hexadecimal: A base 16 numbering system used to represent binary values. Hex uses the numbers 0-9 and the letters A-F: usually notated by an "h" (for example, "4CF h", read "four charley fox, hex"). The result is that one hex digit represents a 4-bit value.

Implicit congestion management: A method of informing the terminal that the network is busy. This method relies on the end-system protocol to detect and fix the congestion problem. TCP/IP is an example of a protocol using only implicit congestion management. See also "explicit congestion management".

In-band: Refers to the type of signalling over the conversion path on an ISDN call. Contrast "out-of-band".

Insufficient Ones: A T1 error condition that is logged when less than one 1 in 16 0s or less than 12.5 % average 1s density is received.

Inter Exchange Carrier (IEC): The long distance company (LE) whose central office provides the point of reference for T1 access. Any common carrier authorized by the FCC to carry customer transmissions between LATAs.

Internet: Refers to the computer network of many millions of university, government, and private users around the world. Each user has a unique Internet Address.

Internet Address (IP Address): A unique 32-bit address for a specific TCP/IP host on a network. Normally printed in dotted decimal format (for example, 129.128.44.227).

Internet Protocol (IP): A protocol used to route data from its source to its destination in an Internet environment. The Internet Protocol was designed to connect to local area networks. Although there are many protocols that do this, IP refers to the global system of interconnecting computers. It is a highly distributed protocol; each machine only worries about sending data to the next step in the route.

Internetwork Packet Exchange (IPX): A NetWare communications protocol used to route messages from one node to another. IPX packets include network addresses and can be routed from one network to another. An IPX packet can occasionally get lost when crossing networks, thus IPX does not guarantee delivery of a complete message. Either the application has to provide that control, or NetWare's SPX protocol must be used.

Interoperable: Devices from different vendors that can exchange information using a standard's base protocol.

I/O Addresses: Locations within the I/O address space of your computer used by a device, such as an expansion card, a serial port, or an internal modem. The address is used for communication between software and a device.

IRQ Level (Interrupt Request Level): The notification a processor receives when another portion of the computer's hardware requires its attention. IRQs are numbered so that the device issuing the IRQ can be identified, and so IRQs can be prioritized.

ISA (Industry Standards Architecture) (pronounced "ice a"): The classic 8 or 16-bit architecture introduced with IBM's PC-AT computer.

ISDN (Integrated Services Digital Network): An International telecommunications standard for transmitting voice, video and data over a digital communications line. ISDN is a world-wide telecommunications service that uses digital transmission and switching technology to support voice and digital data communications. Frame relay was partially based on ISDN's data link layer protocol (LAPD). Frame relay can be used to transmit across ISDN services offering circuit-switched connection at 64K bps and higher speeds. Contrast Public Switched Telephone Network (PSTN).

ITU-TSS (formerly CCITT): International Telecommunications Union-Telecommunications Sector; the United Nations organization that prepares standards ("Recommendations") for resolving communications issues and problems.

K

Key Telephone System (KTS): Phone devices with multiple buttons that let you select incoming or outgoing CO phone lines directly. Similar in operation to a PBX, except a KTS you don't have to dial a "9" for a call outside the building.

Key Service Unit (KSU): A small device containing the switching electronics for a business key telephone system (KTS).

Key Set: A telephone set with several buttons for call holding, line pickup, intercom, autodialing, and so on. Also called a touchtone phone (Ericsson) and a KTS (Key Telephone Set).

L

LAPB: Link Access Procedure Balanced; based on the X.25 Layer 2 specification. A full-duplex point-to-point bit-synchronous protocol commonly used as a data link control protocol to interface X.25 DTEs. LAPB is the link initialization procedure that establishes and maintains communications between the DTE and the DCE.

LAPD: Link Access Protocol for the D-Channel; based on the ISDN Q.921 specification. A full-duplex point-to-point bit-synchronous link-level protocol for ISDN connections; different from LAPB in its framing sequence. Transmission is in units called "frames", and a frame may contain one or more X.25 packets.

Line Coding: The representation of 1s and 0s on a T1 line. The two methods of line coding commonly used, B8ZS and AMI, differ in the restrictions placed on user data. T1 line coding ensures that sufficient timing information is sent with the digital signal to ensure recovery of all the bits at the far end. Timing information on the T1 line is included in the form of 1s in the data stream; a long string of 0s in the data stream could cause problems recovering the data.

Line Termination (LT): The electronics at the ISDN network side of the user/network interface that complements the NT1 at the user side. The LT and the NT1 together provide the high-speed digital line signals required for BRI access.

Listed Directory Number (LDN): The main number assigned by the telco; the number listed in the telephone directory and also provided by Directory Assistance. Some devices can have more than one LDN, such as ISDN devices that have one LDN for voice and another LDN for data.

Local Area Network (LAN): 1) A computer network located on a user's premises within a limited geographical area. Communication within a local area network is not subject to external regulations; however, communication across the LAN boundary may be subject to some form of regulation. 2) A LAN does not use store and forward techniques. 3) A network in which a set of devices are connected to one another for a communication and that can be connected to a larger network.

Local Access and Transport Area (LATA): A post-divestiture geographical area generally equivalent to a Standard Metropolitan Statistical Area. At divestiture, the territory served by the Bell system was divided into approximately 161 LATAs. The Bell Operating Companies (BOCs) provide Intra-LATA services.

Local Exchange Carrier (LEC): The local phone company which provides local (not long distance) transmission services. AKA "telco". LECs provide T1 or FT1 access to LDCs unless the T1 circuit is completely intra-LATA. Inter-LATA T1 circuits are made up of a combination of Access and Long Haul facilities.

Local Management Interface (LMI): A specification for frame relay equipment that defines status information exchange.

Local Loop: A transmission path, typically twisted-pair wire, between an individual subscriber and the nearest public telecommunications network switching center. The wires provide ISDN service, but require an NT1 at the user end and an LT at the network end. AKA, "loop" or "subscriber loop".

Logical Link Control (LLC2): In a local area network, the protocol that governs the exchange of transmission frames between data stations independently of how the transmission medium is shared. The LLC2 protocol was developed by the IEEE 802 commitee and is common to all LAN standards.

Logical Unit (LU): A type of network accessible unit that enables end users to gain access to network resources and communicate with each other.

Long Haul: The T1 element that connects to the Access portion of the long distance company's (LDC's) central office. The LDC is commonly called the point of presence (POP). Each LDC has a number of POPs, located throughout the country. The LDC is also called an IEC (Inter Exchange Carrier).

Management Information Base (MIB): A database of network management information used by the Common Management Information Protocol (CMIP) and the Simple Network Management Protocol (SNMP).

Megacom: An AT&T service with a normal WATS line (typically T1) between the customer premise and the AT&T serving class 4 CO are the customer's responsibility.

MegaLink: BellSouth's leased T1 service.

Message: Associated with such terms as packet, frame, and segment. 1) In information theory, an ordered series of characters intended to convey information. 2) An assembly of characters and sometimes control codes that is transferred as an entry from an originator to one or more recipients.

Modem: A communications device that enables a computer to transmit information over a telephone line. It converts the computer's digital signals into analog signals to send over a telephone line and converts them back to digital signals at the receiving end. Modems can be internal and fit into an expansion slot, or external and connect to a serial port.

Multiplexer (Mux): 1) A device that takes several input signals and combines them into a single output signal in such a manner that each of the input signals can be recovered. 2) A device capable of interleaving the events of two or more activities or capable of distributing the events of an interleaved sequence to the respective activities. 3) Putting multiple signals on a single channel.

Multiprotocol: A device that can interoperate with devices utilizing different network protocols.

Multithreading: The ability of a software system to be able to handle more than one transaction concurrently. This is contrasted to the case where a single transaction is accepted and completely processed before the next transaction processing is started.

N

Nailed Connection: A permanent or dedicated circuit of a previously switched circuit or circuits.

Nailed-up Circuit: A semipermanent circuit established through a circuit-switching facility for point-to-point connectivity.

NAK (Negative Acknowledgment): Communications code used to indicate that a message was not properly received, or that a terminal does not wish to transmit. Contrast with ACK.

Network: A group of computers connected by cables or other means and using software that enables them to share equipment, such as printers and disk drives to exchange information.

Node: Any point within a network which has been assigned an address.

O

Object-Orientated: A method for structuring programs as hierarchically organized classes describing the data and operations of objects that may interact with other objects.

Office Channel Unit - Data Port (OCU-DP): The CO channel bank used as the interface between the customer's DSU and the channel bank.

Off-hook: The condition of a device that has accessed a phone line, with or without using the line. In modem use, this is equivalent to a telephone handset being picked up. Dialing and transmission are allowed, but incoming calls are not answered. Contrast "on-hook".

Off Premise Extension (OPX): An extension or phone that terminates in a location other than that of the PBX. Commonly used to provide a corporate member with an extension of the PBX at home.

Ones Density: The measure of the number of logical 1s on a T1 line compared to a given total number of bits on that line; used for timing information in data recovery in AMI and B8ZS.

On-Hook: The condition of a device which has not accessed a phone line. In modem use, this is equivalent to a telephone handset that has not been picked up. In other words, it can receive an incoming call. Contrast "off-hook".

Open Shortest Path First (OSPF): A hierarchical Interior Gateway Protocol (IGP) routing algorithm for IP that is a proposed standard for Internet. OSPF incorporates least-cost routing, equal-cost routing, and load balancing.

Outage: The measure of the time during which a circuit is not available for use due to service interrupt. Outage is the complement of circuit "availability" (100% minus % available = % outage).

Out-of-band: Signaling that is separated from the channel carrying the information (the voice/data/video signal is separate from the carrier signal). Dialing and various other "supervisory" signals are included in the signaling element. Contrast "In-band" signaling.

Out of Frame (OOF): A T1 alarm condition that is logged on the loss of 2, 3, or 4 of 5 consecutive FT framing bits.

P

Packet: 1) In data communication, a sequence of binary digits, including data and control signals, that is transmitted and switched as a composite whole. The data, control signals and, possibly, error control information are arranged in a specific format. 2) Synonymous with data frame. 3) In TCP/IP, the unit of data passed across the interface between the Internet layer and the link layer. A packet includes an IP header and data. A packet can be a complete IP datagram or a fragment of an IP diagram. 4) In X.25, a data transmission information unit. A group of data and control characters, transferred as a unit, determined by the process of transmission. Commonly used data field lengths in packets are 128 or 256 bytes. 5) The field structure and format defined in the CCITT X.25 recommendation.

Packet Assembler/Dissembler (PAD): Used by devices to communicate over X.25 networks by building or stripping X.25 information on or from a packet.

Packet Data: The information format ("packetized") used for packet-mode calls.

Packet Mode: Refers to the switching of chunks of information for different users using statistical multiplexing to send them over the same transmission facility.

Parity bit: An extra bit attached to each byte of synchronous data used to detect errors in transmission.

Permanent Virtual Circuit (PVC): A connection between two endpoints dedicated to a single user. IN ISDN, PVCs are established by network administration and are held for as long as the user subscribes to the service.

Physical Unit (PU): The component that manages and monitors the resources (such as attached links and adjacent link stations) associated with a node, as requested by an SSCP via an SSCP-PU session. An SSCP activates a session with the physical unit in order to indirectly manage, through the PU, resources of the node such as attached links. This term applies to type 2.0, type 4, and type 5 nodes only.

Point of Presence (POP): The central office's end points of the long distance carriers.

Point to Point Protocol (PPP): A protocol that lets a PC user access TCP/IP (Internet member) using an ISDN terminal adapter or a high-speed modem over a standard telephone line.

Port: A location for input or output data exchange. Computers, muxes, and so on have ports for various purposes.

Primary Rate Interface (PRI): Used on ISDN. In North America, and Japan, PRI is one 64Kbps D channel and 23 B channels. Elsewhere, it is one D channel and 30 B channels.

Primitive: An abstract representation of interaction across the access points indicating that information is being passed between the service user and the service provider. The OSI Reference Model defines four types of primitives: Request, Indication, Response and Confirm.

Private Branch Exchange (PBX): A telephone exchange located on the customer's premises. The PBX provides a circuit switching facility for telephone extension lines within the building, and access to the public telephone network. See also "Exchange".

PROM (Programmable Read Only Memory - pronounced "prom"): A permanent memory chip that can be programmed or filled by the customer after by the manufacturer has set initial values. Contrast with ROM.

Protocol: 1) A set of semantic and syntactic rules that determines the behavior of functional units in achieving communication. 2) In Open Systems Interconnection architecture, a set of semantic and syntactic rules that determine the behavior of entities in the same layer in performing communication functions. 3) In SNA, the meanings of and the sequencing rules for requests and responses used for managing the network, transferring data, and synchronizing the states of network components. 4) Synonymous with line control discipline.

PSTN (Public Switched Telephone Network): A worldwide public voice telephone network that is used as a telecommunications medium for the transmission of voice, data and other information.

Public Data Network (PDN): A packet-switched network that is available to the public for individual ("subscriber") use. Typically, controlled by a government or a national monopoly.

Public Switched Telephone Network (PSTN): The group of circuit-switching voice carriers, which are commonly used as analog data communications services.

Pulse Code Modulation (PCM): 1) In data communication, variation of a digital signal to represent information; for example, by means of pulse amplitude modulation (PAM), pulse duration modulation (PDM), or pulse position modulation (PPM). 2) Transmissions of analog information in digital form through sampling and encoding the samples with a fixed number of bits.

Pulse dialing: One of two methods of dialing a telephone, usually associated with rotary-dial phones. Compare with "tone dialing".

Q

Quantizing: The process of analog-to- digital conversion by assigning a range, from the contiguous analog values, to a discrete number.

R

Random Access Memory (RAM): A computer's primary workspace. All data must be stored in RAM (even for a short while), before software can use the processor to manipulate the data. Before a PC can do anything useful it must move programs from disk to RAM. When you turn it off, all information in RAM is lost.

Rate Enforcement: The concept in frame relay where frames sent faster than the CIR are to be carried only if the bandwidth is available, otherwise they are to be discarded. The frame relay network assumes that anything exceeding the CIR is of low priority. Rate enforcement makes sure that the network will not get so congested that it isn't able to meet the agreed on CIR.

Recognized Private Operating Agency (RPOA): A corporation, private or government-controlled, that provides telecommunications services. RPOAs, such as AT&T, participate as non-voting members in the CCITT.

Red Alarm: A T1 error condition generated when a local failure (e.g., loss of synchronization) iexists for 2.5 seconds, causing a Carrier Group Alarm (CGA). See also "Blue Alarm" and "Yellow Alarm".

Request for Comment (RFC): A set of papers in which Internet standards (published and proposed), along with generally-accepted ideas, proposals, research results, and so on are published.

Ring Down Box: A device that emulates a CO by generating POTS calls for testing and product demos.

Ring Down Circuit: A tie line connecting phones where picking up one phone automatically rings another phone. A feature used for emergencies to alert the person at the other phone of the incoming call.

RJ-11: An industry standard interface used for connecting a telephone to a modular wall outlet; comes in 4-and 6-wire packages.

RJ-45: An 8-wire modular connector for voice and data circuits.

Robbed Bit Signaling: The popular T1 signaling mechanism where the A and B bits are sent by each side of the T1 termination and are "buried" in the voice data of each voice channel in the T1 circuit. Since the bits are "robbed" infrequently, voice quality is remains relatively uncompromised. See "bit robbing".

The robbed-bit signaling technique is used in D4 channel banks to convey signaling information. The eighth (least significant) bit of each of the 24 8-bit time slots is "robbed" every sixth frame to convey voice-related signaling information such as on-hook, and off-hook, for each channel.

Router: A device that connects two networks using the same networking protocol. It operates at the Network Layer (Layer 3) of the OSI model for forwarding decisions.

Routing Information Protocol (RIP): A distance vector-based protocol that provides a measure of distance, or hops, from a transmitting workstation to a receiving workstation.

RS232-C: An EIA standard for a serial interface between computers and peripheral devices (modem, mouse, etc.). It uses a 25-pin DB-25, or a 9-pin DB-9 connector. The RS-232 standard defines the purposes, electrical characteristics and timing of the signals for each of the 25 lines.

RS-422: The EIA standard for a balanced interface with no accompanying physical connector. RS-422 products can use screw terminals, DB-9, various DB-25, and DB-37 connectors.

RS-530: The EIA standard for the mechanical/electrical interface between DCEs and DTEs transmitting synchronous or asynchronous serial binary data. RS-530 provides for high data rates with the same connector used for RS-232; however, it is incompatible with RS-232.

S

Serial Port: The connector on a PC used to attach serial devices (those that need to receive data one bit after another), such as a mouse, a printer or a modem. This consists of a 9- or 25-pin connector that sends data in sequence (bit by bit). Serial ports are referred to as "COMx" ports, where x is 1 to 4 (COM1 through COM4). A serial port contains a conversion chip called a "UART" which translates between internal parallel and external serial formats.

Service: The requirements offered by an RPOA to its customers to satisfy specific telecommunications needs.

Severely Errored Seconds (SES): Refers to a typical T1 error event where an error burst occurs (a short term, high bit-error rate that is self-clearing). Per the ITU-T (CCITT) G.821: any second in which the BER is less than $1x10^{-3}$.

Signaling: The process of establishing, maintaining, accounting for, and terminating a connection between two endpoints. For example, the user premises and the telco CO. Central office signals to the user premises can include ringing, dial tone, speech signals, and so on. Signals from the user's telephone can include off-hook, dialing, speech to far-end party, and on-hook signals. In-band signaling techniques include pulse and tone dialing. With common channel signaling, information is carried out-of-band.

Simple Network Management Protocol (SNMP): TCP/IP protocol that allows network management.

Simultaneous Voice Data (SVD): A technology for letting a user send data via a modem, and use a handset to talk to another user at the same time over the same connection. The alternative, making a second call, can be expensive or even impossible. The uses for SVD are telecommuting, videoconferencing, distant learning, tech support, etc.

Stop Bit: One of the variables used for timing in asynchronous data transmission. Depending on the devices, each character may be trailed by 1, 1.5, or 2 stop bits.

Superframe (D4): A T1 transmission format that consists of 12 DS1 frames, or 2316 bits. A DS1 frame consists of 193 bit positions. A frame overhead bit is in the first position, and it is used for frame and signaling phase alignment only.

Subscriber Loop: See "Local loop".

Switched 56: A circuit-switched (full duplex digital synchronous data transmission) service that lets you dial a number and transmit data to it at 56K bps. It is a relatively low cost service, widely used in North America for telecommuting, videoconferencing and high speed data transfers. Many phone companies are (or will be) phasing out Switched 56 in favor of ISDN service.

Switched Virtual Circuit (SVC): A type of data transmission where the connection is maintained only until the call is cleared.

Switched Line: In communications, a physical channel established by dynamically connecting one or more discreet segments. This connection lasts for the duration of the call after which each segment may be used as part of a different channel. Contrast with leased line.

Switched Network: A network in which a temporary connection is established from one point via one or more segments.

Synchronous Data Link Control (SDLC): A discipline conforming to subsets of the Advanced Data Communications Control Procedures (ADCCP) of the American National Standards Institute (ANSI) and High-level Data Link Control (HDLC) of the International Organization for Standardization, for managing synchronous, code-transparent, serial-by-bit information transfer over a link connection. Transmission exchanges may be duplex, or half-duplex over switched or nonswitched links. The configuration of the link connection may be point-to-point, multipoint, or loop.

Synchronous Transmission: The transmission of data which involves sending a group of characters in a packet. This is a common method of transmission between computers on a network or between modems. One or more synchronous characters are transmitted to confirm clocking before each packet of data is transmitted. Compare to Asynchronous Transmission.

Systems Network Architecture (SNA): The description of the logical structure, formats, protocols, and operational sequences for transmitting information units through, and controlling the configuration and operation of of, networks.

T

Tariff: The rate/availability schedule for telephone and ISDN services from a regulated service provider.

TCP/IP: A set of communication protocols that support peer-to-peer connectivity functions for both local and wide area networks.

T Carrier: The generic name for a digitally multiplexed carrier system. In the North American digital hierarchy, a T is used to designate a DS (digital signal) level hierarchy. Examples: T1 (DS1) is a 1.544 M bps 24-channel designation. In Europe, T1 is called E1. The T Carrier system was originally designed for transmitting digitized voice signals, but has since been adapted for digital data applications.

T1: A digital transmission link capable of 1.544M bps. T1 uses two pairs of normal UTP, and can handle 24 voice conversations, each digitized at 64K bps. T1 is a standard for digital transmission in the U.S., Canada, Japan and Hong Kong. T1 is the access method for high-speed services such as ATM, frame relay, and SMDS. See also T Carrier, T1 line and FT1.

T1 Channel Tests: A set of diagnostics that vary by carrier, used to verify a T1 channel operation. Can include Tone, Noise Level, Impulse Noise Level, Echo Cancellors, Gain, and Crosstalk testing.

T1 Framing: To digitize and encode analog voice signals requires 8000 samples per second (twice the highest voice frequency of 4000 Hz). Encoding in an 8-bit word provides the basic T1 block of 64K bps for voice transmission. This "Level 0 Signal, as its called, is represented by "DS-0", or Digital Signal at Level 0. 24 of these voice channels are combined into a serial bit stream (using TDM), on a frame-by-frame basis. A frame is a sample of all 24 channels; so adding in a framing bit gives a block of 193 bits (24x8+1=193). Frames are transmitted at 8000 per second (the required sample rate), creating a 1.544M (8000x193=1.544M) transmission rate.

T1 Line: A digital communications facility that functions as a 24-channel pathway for data or voice. A T1 line is composed of two separate elements: the Access element and the Long Haul element.

T1 Mux: A device used to carry many sources of data on a T1 line. The T1 mux assigns each data source to distinct DS0 time slots within the T1 signal. Wide bandwidth signals take more than one time slot. Normal voice traffic or 56/64K bps data channels take one time slot. The T1 mux may use an internal or external T1 DSU; a "channel bank" device typically uses an external T1 CSU.

Transmission Control Protocol / Internet Program (TCP/IP): A multilayer set of protocols developed by the US Department of Defense to link dissimilar computers across dissimilar and unreliable LANs.

Terminal: The screen and keyboard device used in a mainframe environment for interactive data entry. Terminals have no "box", which is to say they have no file storage or processing capabilities.

Terminal Adapter (TA): An ISDN DTE device for connecting a non-ISDN terminal device to the ISDN network. Similar to a protocol converter or an interface converter, a TA connects a non-ISDN device between the R and S interfaces. Typically a PC card.

Tie line: A dedicated circuit linking two points without having to dial a phone number. The line may be accessed by lifting the telephone handset or by pushing a button.

Time-Division Multiplexing (TDM): Division of a transmission facility into two or more channels by allotting the common channel to several different information channels, one at a time.

Time Slot: One of 24 channels within a T1 line. Each channel has a 64K bps maximum bandwidth. "Time slot" implies the time division multiplexing organization of the T1 signal.

Toll Call: A call to a location outside of your local service area (a long distance call).

Tone dialing: One of two methods of dialing a telephone, usually associated with Touch-Tone® (push button) phones. Compare with pulse dialing.

Topology: Physical layout of network components (cables, stations, gateways, and hubs). Three basic interconnection topologies are star, ring, and bus networks.

Transmission Control Protocol (TCP): A communications protocol used in Internet and in any network that follows the US Department of Defense standards for internetwork protocol. TCP provides a reliable host-to-host protocol between hosts in packet-switched communications networks and in interconnected systems of such networks. It assumes that the Internet protocol is the underlying protocol.

Transport Layer: Layer 4 of the Open Systems Interconnection (OSI) model; provides reliable, end-to-end delivery of data, and detects transmission sequential errors.

Transport Protocol Data Unit (TPDU): A transport header, which is added to every message, contains destination and source addressing information that allows the end-to-end routing of messages in multi-layer NAC networks of high complexity. They are automatically added to messages as they enter the network and can be stripped off before being passed to the host or another device that does not support TPDU's.

Trunk: Transmission links that interconnect switching offices.

TSR (terminate and stay resident): A software program that remains active and in memory after its user interface is closed. Similar to a daemon in UNIX environments.

Tunneling: Encapsulation data in an IP packet for transport across the internet.

Twisted pair wiring: A type of cabling with one or more pairs of insulated wires wrapped around each other. An inexpensive wiring method used for LAN and telephone applications, also called UTP wiring.

U

UART (Universal Asynchronous Receiver/Transmitter) (pronounced "you art"): A chip that transmits and receives data on the serial port. It converts bytes into serial bits for transmission, and vice versa, and generates and strips the start and stop bits appended to each character.

UNIX: An operating system developed by Bell Laboratories that features multiprogramming in a multi-user environment.

Unshielded Twisted Pair (UTP): Telephone-type wiring. Transmission media for 10BASE-T.

V

V.25bis: An ITU-T standard for synchronous communications between a mainframe or host and a modem using HDLC or other character-oriented protocol.

V.54: The ITU-T standard for local and remote loopback tests in modems, DCEs and DTEs. The four basic tests are:
- local digital loopback (tests DTE send and receive circuits),
- local analog loopback (tests local modem operation),
- remote analog loopback (tests comm link to the remote modem), and
- remote digital loopback (tests remote modem operation).

Virtual Circuit: A logical connection. Used in packet switching wherein a logical connection is established between two devices at the start of transmission. All information packets follow the same route and arrive in sequence, but do not necessarily carry a complete address.

Index

A

About the Internet 96
Accessories, ordering 95
Application, typical 8
Applications setup, changing 75
Archie 99
Auto Call 63
Automatic disconnection, billing/security 63

B

Back Panel 14
Billing options 64
Billing/Misc default setup, Master 31, 51
Billing/Misc parameters, changing 63
Boot LED 13

C

Cabling the MultiVOIP
  Local site 20
  Remote site 54
Call progress, viewing 74, 80
Canadian limitations notice 105
Channel interface default setup, Master 27, 47
Channel interface default setup, Slave 47
Channel interface parameters, changing 60
Channel totals, viewing 81
Coder, voice/fax 62
Collision 13
Command port cable, diagram 101
Command port connector 14
Configuration port setup 58
Configuration utilities, overview 58
Configuring the MultiVOIP, overview 18
Configuring your Slave MultiVOIPs 46
Connectors
  Command 14
  Power 14
Current loss 61
Custom tone pair settings 65

D

Deploying the VOIP network
  Remote site administrator 54
  VOIP administrator 54
Deploying the VOIP network, overview 18
DNS 99
Download firmware 58
Downloading firmware 82
DTMF gain, voice/fax 62
Dynamic jitter buffer 64

E

E&M connector 14
E&M interface 61
E&M jumper block positioning 21
EMC, safety and terminal directive compliance 105
Ethernet connector 14
Ethernet LEDs 13

F

Fax branding statement 103
FCC Part 15 statement 103
FCC Part 68 Telecom statement 104
Finger 99
Firewall, installing with MultiVOIPs 24
Flash Hook Timer 60
Front Panel, description 13
FTP (File Transfer Protocol) 99
FXO connector 14
FXO disconnect on 61
FXO interface 61
FXS connector 14
FXS interface 61

G

Gatekeeper
  Overview 6
Gopher 99

H

H.323 protocol
  Overview 6
HTML 99
HTTP 99

I

Industry Canada statement 103
Installing MultiVOIP software 24
Installing software, overview 18
Inter digit time 60
Internet
  Multi-Tech's Web site 96
Internet protocol (IP) 100
IP default setup, Master 27
IP default setup, Slave 46
IP parameters, changing 73
IP statistics 78

J

Jitter buffer, dynamic 64
Jitter value 63

L

LAN cable, diagram 101

LAN-based remote configuration 87
LEDs, front panel 13
Limited warranty 94
Log entries, reports 81
Log entry details, viewing 80

M

Max Baud rate, voice/fax 63
Modem-based remote configuration 85
MultiVOIP 400/800
  Accessories, ordering 95
  Back panel, description 14
  Cabling diagrams 101
  Cabling procedure 20
  Deploying the VOIP network 54
  E&M positioning procedure 21
  Firewall, installing with MVP400/800 24
  Front panel, description 13
  Installing the MultiVOIP software 24
  Introduction 6
  Service 95
  Technical specifications 15
  Technical support 96
  Typical application 8
  Unpacking 19
  Upgrade procedure 82
MultiVOIP Main menu, introduction 58
MultiVOIP Main menu, screen 59
MultiVOIP, Telnet server menu 89
MVP200
  Configuring the Slave MultiVOIP 46

O

Optimization factor 64
Ordering accessories 95

P

Packets lost/received 74
Password, SNMP 75
Phone directory database, changing 66
Ping 99
POP 99
Power
  Connector 14
Power LED 13
Pulse generation ratio 65

R

Receive Data 13
Regeneration 60
Regional default setup, Master 31, 51
Regional parameters, changing 65
Regulatory information 103
Remote configuration
  LAN-based procedure 87
  Modem-based procedure 85
Remote management 88
  Telnet 88
  WEB management 90
Remote site administrator, deployment 54
Remote site cable connection 54
Reports, log entries 81
Ring count, FXO and FXS 61
Ringer equivalence number 105

S

Safety Warning
  Battery and port interfaces 19
  Telecom 18
Service 95
Silence detection 61
Slave MultiVOIP, configuring 46
Slave MultiVOIPs, configuring 46
Slave status 69
SMTP 99
SNMP 99
SNMP agent, enabling 75
SNMP statistics 79
Software 58
  Description 58
Specifications
  Command Port 15
  Electrical/Physical 15
  Ethernet Port 15
Station identification 68
Statistics, viewing
  IP 78
  SNMP 79

T

TCP packets 78
TCP/IP 88, 98
Technical support 96
Telnet 88, 99
  Proxy Server Configuration 90
Telnet server menu 89
Tone detection 61
Tone pairs, customizing 65
Traceroute 99
Transmit Data 13
Typical application 8

U

UDP packets 78
Uninstall ProxyServer configuration 58
Unpacking your MultiVOIP 19
Upgrading software version 82

V

Viewing statistics 78

Voice coder, voice/fax 62
Voice over IP configuration..... 89
Voice/fax channel connectors 14
Voice/Fax channel connectors, diagram 101
Voice/Fax channel LEDs 13
Voice/Fax default setup, Master 29, 48
Voice/Fax parameters, changing 62

W

Warranty 94
WEB Server 90